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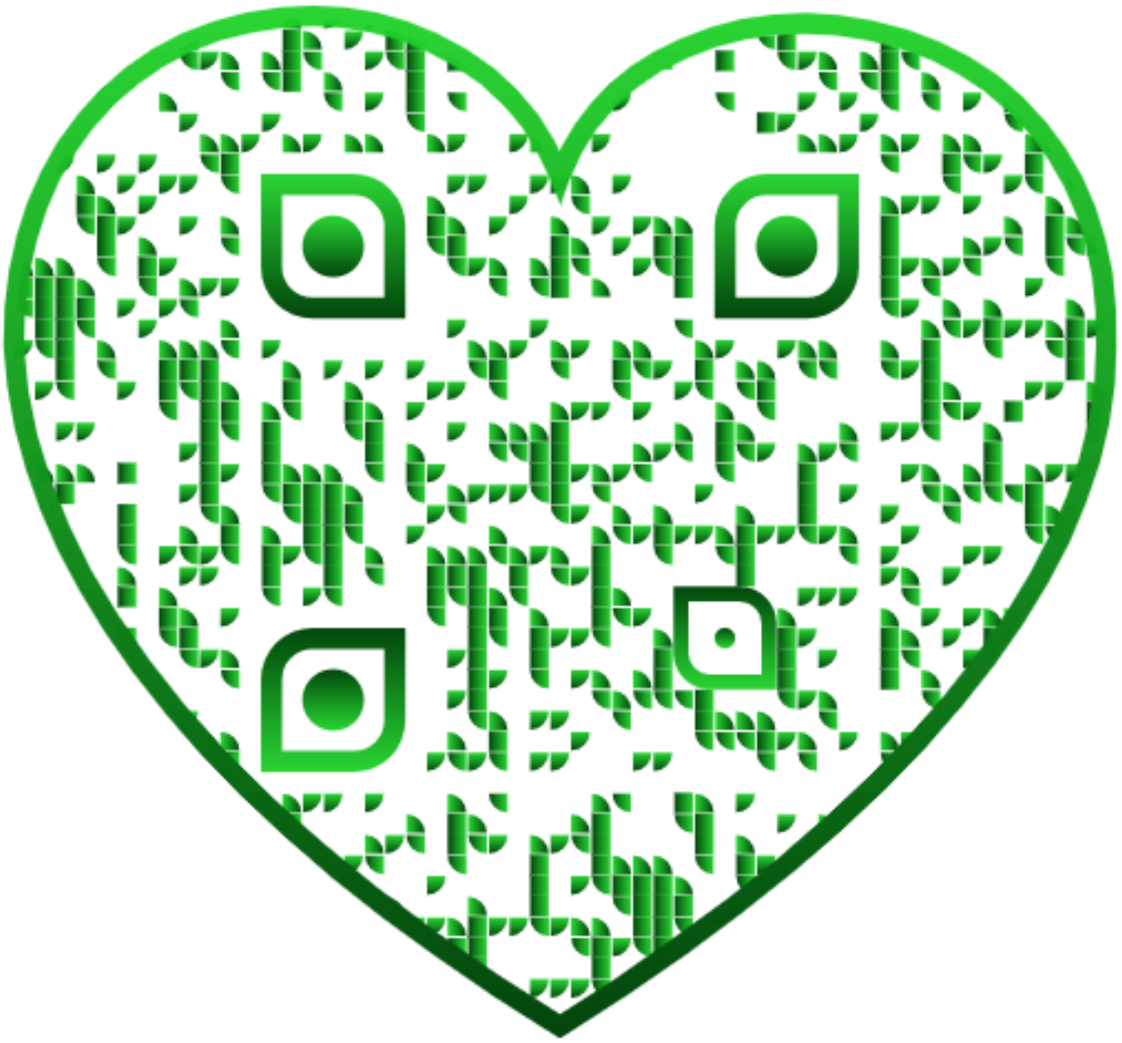

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KEEPING IN TOUCH

As this month's cover feature shows, U2's residency at the new Vegas Sphere is revolutionary, both from a technical and an artistic point of view. In its ambition, scale and sheer novelty, it also highlights the growing gulf between the live and studio worlds.

For years, the studio was the heart of sonic innovation. As artists explored multitracking, overdubbing, editing, sequencing, sampling and synthesis, recording studios were constantly pushed into finding solutions to new problems. But today, live sound is driving the technological minibus, while a sort of museum mentality seems to pervade the studio world. Which of us can truthfully say we aren't swayed more by the lure of a vintage console, or a collection of classic keyboards, than by the latest AI breakthroughs or haptic control devices? How many of the new products reviewed in this magazine sell themselves through the promise of recapturing some sort of old magic?

If we're honest, moreover, I think most of us would have to admit that this isn't always about the pursuit of the best possible sound. The average listener will neither know nor care whether we mixed a track using expensive outboard or cheap plug-ins. Collecting vintage equipment is as much about vibe as it is about the results. It's selling something that artists and producers can't get from their laptops. But does a studio have to hark back to the golden age of recording to do this? Why shouldn't cutting-edge technology also be an effective tool for selling studio time? Why do clients get excited about old valve mics and not about, say, a Soundfield mic, something that really does open up new possibilities?

If there's one new technology that can act as a selling point for studios right now, it's probably immersive audio, and those who have dived into Atmos seem to be reaping the rewards. At the same time, though, there's a lot of scepticism among those who haven't. That's understandable, given the often underwhelming experience of spatial audio on headphones and the steep learning curve associated with Atmos mixing. But when scepticism drifts into cynicism, it raises questions about the future of the recording studio. If we're not going to embrace new technologies, if we're not willing to learn and exploit new tools, what are studios going to be for? When a new generation of artists and producers no longer cares about the equipment that was used to record the Beatles, what reason can we give them to hire us?

Abbey Road has a past like no other studio, but it's thriving today because it has invested in the future. For studios without history on their side, staying relevant is even more important.

Sam Inglis
Editor In Chief

"If we're not willing to learn and exploit new tools, what are studios going to be for?"

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JOHN WALDEN

Baby Audio Transit

Multi-effects Plug-in

Transitions are the name of the game here, and as always Baby Audio and Andrew Huang encourage you to get creative!

YouTube has become a promising route for musicians, songwriters and producers to find an audience for their creative output, and there have been some amazing success stories, one of which is Canadian musician/producer Andrew Huang. At the time of writing, His YouTube channel has well over 2,000,000 subscribers, who are attracted by the eclectic combination of weird and wonderful original music, discussions on music production, and quirky (but always interesting) musical challenges and collaborations.

His latest collaboration is with the plug-in development team at Baby Audio. Both parties are well known for a creative approach to audio processing, so it's perhaps not surprising that the result, Transit, is a multi-effects plug-in that's designed to spice up your musical transitions. But it's one with a bit of a twist — I mean that very literally, given the large 'T' knob on the GUI!

Plug & Play

By their nature, multi-effects plug-ins can be complex affairs, and considered against that context Transit's GUI is something of a triumph! I'll come back to the T macro knob in a minute, but you can place one of the 18 different effect types into each of the seven other slots in any order you choose, the only limitation being that only one instance of each effect can be used per preset. The top strip provides access to a very well-stocked preset browser and some useful randomisation and MIDI options, while at the bottom you have horizontal sliders to control the overall Mix and Output levels.

The controls available in each of the effects panels vary by effect but, taking the Filter 24 module as an example, you get four rotary knobs allowing you to configure filter Type, Cutoff, Resonance and Mix levels. The top strip of each module slot includes a bypass button, a drag-and-drop button for changing the module's position in the signal chain, the option to load/save module-specific presets and the randomisation button for the module. The small dots in the centre allow you to link particular controls to the main T macro control; the dot turns yellow when linked in this way and you can then specify a minimum and maximum range for that specific effect parameter. As you then move the T knob, the linked parameters sweep through the specified range. You have the option to sweep a control forwards and backwards and to change the linearity of the response (just drag on the straight line in the centre of the specific knob; it can be bent).

Given the plug-in's name and its tagline as a creator of transition effects, it's worth emphasising here that Transit can also operate as a conventional 'static' multi-effects processor. Indeed, given the flexibility and diversity of effects provided here, Transit's potential as a more straightforward creative multi-effects processor really shouldn't be downplayed. You can either set the T control to a fixed position or, on the top strip, toggle the Infinity button to switch the effects chain into an on state, even when the T knob is set to zero. Individual controls linked to the global macro knob then adopt their starting point values, while non-linked controls simply use the set value.

Tools In The Box

The list of effects is impressive. You get panning, bit-crushing, various modulation effects (chorus, flanger, phaser, tremolo), distortion, a multiband compressor (OTT), a pumping effect (which achieves a result rather like a side-chained compressor), reverb, delay, stereo widening, pitch-shift, and two filters (12 and 24 dB filter slopes and each with four filter styles). You also get a very useful Utility module with Gain, Tone and Pan controls, which can be great for some pre- or post-processing at a specific point in your signal chain.

All of these modules process the audio signal you feed through Transit and, where appropriate, modules (eg. Reverb, Delay, Distortion, Chorus) have dedicated Mix controls. But when it comes to transitions, you also might also want to create new sounds, rather than just process the signal. To that end, there are two modules, Noise and Osc, and as the names suggest these provide you with options to add noise or sound from a basic oscillator; the latter offers four waveform options. They can be used independently or together and, with a dollop of ambience, can easily be used to create a whooshing riser, either to blend in »

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Transit's Noise and Osc modules are perfect for adding an additional sonic element to your transitions.

with whatever audio you are feeding into Transit or to be used on its own.

The effects sound good, and while things like the reverb or delay won't offer the detailed control you might find in dedicated plug-ins, they are very usable indeed in a multi-effects chain. The compact control set for each module — there are never more than four controls to adjust — is a sensible design decision too. It helps keep the workflow, er... flowing.

All Rise

Having configured your effects chain and linked the desired parameters to the main T macro knob, applying your effects chain to create a riser, transition or drop effect simply requires you to turn that control: you can use the mouse, your standard DAW options to link to a hardware controller or create automation data. The last of those is both flexible and precise: a single automation lane lets you control the exact length of the transition for all the macro-linked effects parameters, whether you want it to occur over a beat, a bar, or a full song section. Whatever the effects chain you've created, this is a super-efficient means of controlling when it occurs and the depth of any processing applied. Risers, full transitions (increase T, hit a peak, and then decrease T again), or drops (moving T from a downbeat; some pitch-shift can work well here). It's an impressively simple process.

The T knob, though, is not your only option. Clicking the double arrow icon in the T module switches the synchronisation to Sequencer mode. Click (or automate) the Play button and the transition will trigger at the start of the next bar, lasting however many bars you set (up to a maximum of 16). The Pendulum mode takes the transition up and then back down again (so you get a rise/fall effect). Finally, and very cool indeed, once it's triggered the Loop switch keeps the sequencer running all the time. So if you want to use Transit to add a constant sense of subtle (or perhaps not so subtle!) effects-based movement to any sound, it can do that too.

Transit also offers sequencer-style control to trigger your effects modulation, and the choice of effects modules is impressive.

I can only really scratch the surface of what's possible with Transit — this is definitely a plug-in you'll want to spend some time experimenting with — but the randomisation options are particularly worth noting, because they offer you more control over the results than most. You can apply randomisation at the global level (the dice icon in the upper control strip) or at the individual module level (the dice icon found at the top right of each module). There is also a lock/unlock system that lets you control which parameters will be subject to randomisation, and this includes the effects choices themselves. As you can also trigger the global randomisation button via MIDI, if you have something like a synth loop or drum loop being processed by Transit, and you keep the degree of randomisation to just a few key effects controls, you can 'perform' with the effects and gently randomise what they are doing during playback. It's a very cool way to generate some low-level ear candy in an otherwise static track. It does take a little time to work out exactly how all the randomisation options interact but hopefully you can see that they're well worth exploring.

When rolling the dice you can choose which parameters will be affected.

A Multitude of Multis

The bottom line here is that Transit sounds great, provides some fabulous creative sound processing options, and yet still manages to be very accessible. Whether for pop, electronic music, or creative effects for scoring work, it has plenty to offer, and can be equally at home being used on a single instrument, an instrument bus or your master bus. Individually, the effects are all easy to use and the the well-thought-out GUI and control set deliver an overall workflow that encourages experimentation.

There are plenty of other creative multi-effects plug-ins, whether built into DAWs (such as Cubase's FX Modulator) or available separately (Narcotic, ShaperBox 3, Movement, Boost, Tantra 2). But for me Transit's most obvious point of comparison is Sugar Bytes' Turnado. Both are massively creative and sonically impressive, and while I'd say Turnado does probably have the edge in terms of depth, Transit is definitely less intimidating. It's impressive, compelling, a joy to use and very competitively priced (the free, fully functional trial version just adds a few seconds of silence every minute during playback and is well worth giving a try). Recommended!

summary

Transit can create fabulous ear-candy transitions but can also serve as a creative general multi-effects processor. The UI is beautifully designed for ease of use.

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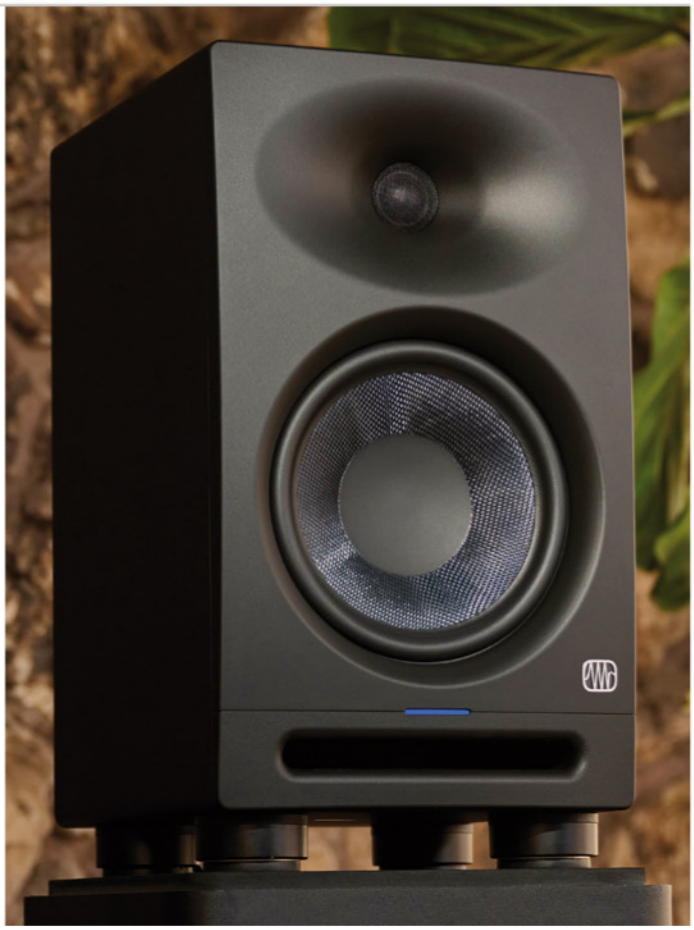

PAUL WHITE

The new Eris Studio series from PreSonus comprises three models of differing sizes, based around four-, five- and eight-inch woofers, and it's the largest of the three we're reviewing here. There's nothing at all unusual about the physical format of them — in fact they follow the most common monitor format found in smaller studios and home facilities, being a two-way active system in a ported cabinet. In order to control the dispersion of the tweeter at the crossover point and to maximise the width of the sweet spot, the tweeter sits in an EBM (Elliptical Boundary Modelled) waveguide, which has a 120 x 60-degree (horizontal x vertical) dispersion. There have been earlier two-way models of a similar format in the Eris range, and they have been well-received at their price point, but this new Studio model has had a significant revamp and features a woven-composite-cone woofer and a 1.25-inch, soft-domed silk tweeter. The design aim was to provide a richer, more tightly controlled low end and detailed highs, which I think it would be fair to say they have achieved. The woven-composite cone construction combines light weight with mechanical stiffness.

These speakers are bi-amped, crossing over at 3.5kHz, with a total of 140 Watts of amplifier power (75W for the lows, 65W for the highs) and cover the 35Hz-20kHz frequency range, though no tolerances are supplied. A maximum SPL of 106dB peak at one metre is quoted.

Internally braced, the cabinets are made from MDF with a black vinyl veneer, with the front-exiting reflex port sited close to the bottom of the baffle. Slightly rounded cabinet edges, speaker aperture and port exit help avoid cabinet-edge diffraction. All the controls and connectors are on the metal rear panel and have the same layout and format as those found on the more costly coaxial Eris Pro 8s. An LED bar on the front of the speaker shows white in standby mode and glows blue when the speaker is active.

The input options are all analogue and offer a choice of unbalanced RCA phono, TRS balanced quarter-inch jack or balanced XLR, with an overall input volume control. Acoustic tuning controls are available to compensate for placement relative to boundaries: an Acoustic Space switch allows for full-, half- and quarter-space placement (LF trim of 0, -2 and -4 dB respectively), plus separate HF and mid controls variable over ±6dB. There's also a high-pass filter with options for off, 80Hz or 100Hz. Protection is built in to filter out RF and provide output-current limiting and over-temperature protection. There's also a soft-start circuit to prevent power-on or -off thumps, as well as a subsonic filter to keep out harmful ultra-low frequencies.

In Use

My tests were carried out in my own studio space, well away from walls and corners, alongside my old faithful Event Opals. Thought the Eris Studio 8s cost around the same as the tax on the Opals, they certainly didn't disgrace themselves. Indeed, by dialling back the midrange control just slightly, I managed to get a pretty good timbral match between the two, though the Opals still won out on depth of bass, with a little more low-end tightness and a better overall sense of focus in the midrange. Nevertheless, the bass end of the Eris Studio 8s is still punchy and sounds tighter than I've heard from many similarly sized monitors in the sub-£1000 bracket, while the highs are clear and detailed without undue harshness thanks to that silk-domed tweeter. The sweet spot is usefully wide, the stereo imaging combines a strong sense of width with a solid centre image, and there's a good sense of depth to the sound stage. Once I'd tamed the slightly forward mids using the rear-panel EQ, the results were most encouraging.

All monitors are a compromise based on cost, the laws of physics and design expertise, but given the affordability of the Eris Studio 8s, I have to say that I am impressed. Their creditable price:performance ratio makes them a good fit for home studios, so if you're in the market for a pair of eight-inch monitors in this price range, then the Eris Studio 8s should definitely be on your shortlist.

summary

The Eris Studio 8s provide a reliable standard of monitoring for home studio owners whose budgets are restricted.

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Universal Audio Lion ’68 Super Lead Amp

Guitar-amp & Speaker Modeller

UA add emulations of three iconic Marshall amps to their acclaimed UAFX amp pedal line-up.

DAVE LOCKWOOD

When UA launched their amp- and speaker-emulation pedals in 2022 with a Vox and two flavours of Fender, there was a notable omission: surely there had to be a Marshall — the iconic sound of rock guitar — in the works? Well, it’s here now, and it is stunningly good! The Lion ’68 Super Lead Amp, to give it its full title, models three flavours of 100W Marshall valve amps: the classic Super Lead, a Super Bass (not just for bass players!) and a Super Lead running on a reduced mains voltage (using a Variac) for extra distortion. Each can be paired with one of six speaker-cab and dual-miking combinations, with a further option of ‘no cab’ for use with an external speaker-sim or, indeed, with a real speaker via a power amp.

The form factor remains the same as for the other amp modellers, with six control knobs and two footswitches. Audio is stereo in and out on unbalanced jacks, and an external PSU capable of at least 400mA is needed; like UA’s other large-format pedals, this is a power-hungry, dual-processor device. Once again, a USB-C connector is used for registering the pedal and firmware updates (you get three additional speaker cabs when you register), and deeper editing and preset handling are only possible using a mobile app that connects over Bluetooth.

I’ve owned and used most of the iconic Marshall models over the years, but ‘the one’ for me, which saw me through countless gigs in the ’70s and ’80s, was a 100W Super Lead, and that just happens to be what most of the Lion ’68 is based on. Not that all 100W Super Leads are the same: they are very much not! Despite being a relatively simple design, the circuit received a number of revisions, components were changed, their values drifted, and indeed their original values varied just enough to make a difference given the component tolerances of the era. By 1968, the Super Lead model featured a massive 5000pF ‘bright’ capacitor as a fixed element (ie. not switchable like on most Fender amps). It’s a big part of what made a Super Lead sound like a Super Lead, provided you could turn the volume up far enough to mitigate its effect — say, somewhere above six, by which time it was very, very loud! Anywhere much below that, it would make the amp sound unusably thin and harsh. It was a common and easy mod, even when these amps were new, to swap out the 5000pF cap for a smaller value, and UA have taken the pragmatic option of pre-modding their virtual Super Lead with a 100pF bright cap, a value used in some earlier Marshalls. Enough, but not too much, though there’s the option also to disable the bright cap altogether in the app. The Super Bass model has no bright cap, and for many players it was a preferable amp for cleaner sounds, or to use with the harsher-sounding fuzz and »

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» distortion pedals of the day. (Jimi did OK with one!)

Input Options

All the Marshalls of this period were non-master-volume amps, of course, but had two Volume controls — one for each channel, marked Volume I and Volume II (high-treble and normal). Each channel also had two input jacks, offering high or low sensitivity. The two inputs on the Lion '68 pedal don't replicate that setup — they are for connecting a stereo input source — but most users will probably be just using a single input to connect a guitar, given that most things that benefit from being stereo can be patched post-amp. The app allows a choice of virtual high- or low-sensitivity inputs, and the pedal itself defaults to replicating a 'Y' connector setup, feeding the guitar into both channels, so you can just turn up whichever one you want, or mix them, which was usually the point of using a 'Y' connector. The app also supports a virtual 'jump lead' option replicating the use of a small link lead between the channels, which has the secondary effect of slightly reducing the input sensitivity. The input routing choice is saved with each preset, and includes the clever 'four-cable' option of the other UAFX amp pedals.

The lower row of knobs features the classic Marshall Bass-Middle-Treble tone control line-up. Located later in the circuit, after two preamp stages, in theory the Marshall tone stack has a harmonically richer signal to work with than a Fender tone stack, but it can often seem to be less effective because so much of the bass has already been removed before the signal reaches it. It was not at all uncommon amongst players of my generation to always turn all the tone controls to 10 on a Super Lead and just fine-tune the top end with the Presence control. The latter is accessed on Lion '68 in the Alt mode that's common to all the UAFX amp pedals: the Middle control becomes Presence, the Bass control determines how much virtual room ambience you have, and the Treble knob gives you a variable boost. What the Boost control actually does changes according to where you set it: just above minimum it adds the slightly smoothed out sound of an Echoplex EP-III preamp; raised a little more it adds clean gain, and further up, you are into mid-boost territory to push the amp into a focused, singing sustain.

OX-type Speaker Modelling

There are six OX-type, dynamically modelled speaker/mic combinations on offer (assuming you've registered to get the extra three): Celestion, JBL and Electro-Voice speakers, in 4x12, 2x12 and 1x12 cab configurations, are matched with a selection of modelled beyerdynamic M 160 and Royer R121 ribbon mics, an AKG C414 capacitor model and Shure SM57 and Sennheiser MD421 dynamic mics. You can't pick and choose your preferred speaker and mic combinations, unlike in the OX software, which will let you pair any and all combinations, but the choices work just fine for me. There always seems to be one cab that's just perfect for whatever tone you've dialled up.

The Lion '68, like all the UAFX amp pedals, has two modes: Live and Preset. In Live mode you are hearing the result of all the current physical controls and app settings, while Preset mode recalls a pre-saved setup, which, unlike on many amp modellers, includes the master Output level control. Although the presets are stored in the pedal, not the app, you can only ever access the most recently loaded one from the pedal itself: to change it you have to use the app. The two footswitches are user-configurable using the app, and can select Live or Preset mode, engage Bypass (buffered), or toggle the boost on and off. Edit mode in the app is essentially unchanged from the original release, so there's still no way to examine the settings used by the presets. If you think of these pedals just as virtual amps and only use Live mode, as I tend to do, this really doesn't matter, but if you want to start from a preset and tweak it, you never know how far each control is from its stored position. By the time you've speculatively moved a couple of them and perhaps got it a bit wrong, the preset no longer sounds like the thing you liked, so there's little point continuing to tweak it. All it takes to improve on this is some way of nulling the controls, such as blinking one of the footswitch LEDs — an indication used in other UA pedals to warn that the control you're adjusting doesn't do anything in the current mode. To store a Live-mode sound you've created, you just hold down the Store switch until the Preset footswitch LED blinks.

The infamous Marshall 'ghost notes' caused by intermodulation between the audio and power supply noise and ripple are fully present and correct, but you can choose to switch them off in the app. I like the effect on distorted leads — it is simply part of the character — but prefer it gone for cleaner, especially chordal playing. There's also a noise gate in the app: effective, if that's something you want, but I can't say I perceived any reasonable settings as noisy enough to use it.

A 'Real' Marshall

Until UA launched their 'one-amp-per-pedal' range, there was probably no modeller made that didn't include some Marshall presets. It's quite easy to do 'generic Marshall': plenty of distortion and a bit too much upper midrange and you can call it done. But Marshalls are more than that. You can't fully know, unless you've played one at sweet-spot volume, quite what the real experience is. Sure, hot-rodded and Variac-ed Marshalls are all about distortion, but most of us didn't hot-rod back in the day, and Marshalls were prized as much as anything for their ability to slide seamlessly between clean and driven tones just by riding a guitar volume control. Marshalls can actually do beautiful cleans and, especially, semi-cleans and, to me, that's what is so special about this pedal.

This isn't 'generic Marshall. This is 'real Marshall'. And as a long-term Marshall user, I recognise so much of what those amps meant to me in the touch response and sonic performance of the Lion '68. This feels like it was a labour of love for the design team, and I'm sure that love will be reciprocated by those who know the real thing, and indeed those now able to experience something of the Marshall magic for the first time.

summary

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McDSP SA-3

Spectral Processing Plug-in

The best plug-ins make under-the-hood sophistication simple in practice...

SAM INGLIS

Based on hardware developed by Hollywood re-recording mixer Mike Minkler, McDSP's SA-2 is a multiband dynamics processor optimised for dialogue. Picking up where that plug-in leaves off, the new SA-3 is dubbed a Spectral Processor rather than a Dialog Processor. This doesn't mean it's ghostly, rather that its applications now go beyond improving the sound of the human voice.

Available in all major native formats on macOS and Windows, and authorised using iLok, SA-3 eclipses its predecessor in terms of the granularity of its processing. Where SA-2 offers five bands of dynamic control, SA-3 is capable of working in over 1000. And as only a sadist would ask the end user to set parameters individually for this many bands, McDSP have developed a friendly, resizeable UI that allows you to direct proceedings with a few clicks.

Suppression Fire

In a nutshell, the SA-3 algorithm identifies resonant frequencies and applies narrow-band gain reduction to suppress them. The Detect knob acts like a global threshold control, and a good starting point is to turn this knob clockwise until you start to see gain reduction activity in the main graphical window. A Monitor button allows you to audition only what's being taken away from the signal.

There are conventional Attack and Release controls, a Range setting that determines the maximum amount of gain reduction, plus controls labelled Focus and Density. The former "alters the spectral processing algorithm to only target more persistent resonant frequencies", while the latter determines the extent to which the harmonic frequencies of a resonance should be suppressed as well as its fundamental.

By default, SA-3 processing is applied across the entire spectrum, but the Bias button engages an EQ-like interface that allows it to be weighted or targeted towards specific areas. This offers a high-pass filter plus four flexible bands which can be given a parametric or shelving response. Applying an EQ 'boost' leads to more processing taking place in the designated part of the frequency spectrum, whilst cutting at a particular frequency makes it less likely that SA-3 will do anything. The Scale and Shift controls dial in vertical and horizontal offsets for the entire Bias curve, which is quicker than trying to adjust all five bands at once.

Smooth Operator

There are conceptual similarities between SA-3 and plug-ins such as Oeksound's Soothe 2 and Soundtheory's Gulfoss, but in practice, it's distinctly and usefully different from those products, and has its own strengths. Regardless of its rebranding as a general-purpose 'spectral processor', SA-3 still shines most of all on vocals. With an almost magical knack for zapping unwanted honks and exaggerated overtones, it tracks changing notes and timbres smoothly and effortlessly. Bump the high-pass filter up to 4kHz or so and it becomes an absolutely superb de-esser. Changing the Attack time can make quite a difference to the amount of processing that takes place, sometimes more so than adjustments to the Detect or Range controls, but I found that I generally got the most natural results by leaving it somewhere in the middle.

As far as vocals are concerned, in fact, SA-3 is pretty much the proverbial 'make it sound better' box. It doesn't audibly compress voices; nor does it make obvious tonal changes unless pushed quite hard. It just makes it sound as though you'd done a better job of recording them in the first place. On other sources, it always keeps the Hippocratic oath, but depending on the instrument, it's occasionally hard to get it to do enough. For example, even when I dialled in a huge Bias boost at 400Hz and cranked the Detect and Range controls, it was reluctant to get too involved with a ringy snare track. By contrast, it works brilliantly for invisibly polishing things like fingerstyle guitar; and on a DI'ed bass part where the fundamental is just a bit too heavy, SA-3 will follow it up and down the fretboard to keep it locked down in a way that no conventional EQ or multiband compressor can.

There is clearly some very clever tech at work under SA-3's hood, but as ever, that's only part of the story. What's just as important is the way McDSP have developed a simple, intuitive front end to allow you to control that tech. It's almost a challenge to get SA-3 to do something that sounds obviously processed or unnatural. Most of the time you won't know it's doing anything at all — until you hit the bypass button.

summary

With advanced multiband processing controlled by a few well-chosen parameters, SA-3 is an intuitive plug-in that will polish your vocals up a treat.

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Futurephonic Rhythmizer Ultra

MIDI Plug-in

Encourage new ideas with some controlled MIDI randomisation.

JOHN WALDEN

Your mileage may vary but, personally, when it comes to the generation of new musical ideas, I'm more than happy to take help wherever it might come from. For MIDI-heads, Futurephonic's Rhythmizer — just updated from v2.1 to the Ultra version with a redesign of the UI and some new features — might be a suitable source of musical inspiration. This 'MIDI note manipulation' plug-in may already be familiar to Live users but it's now also available in a VST/AU format and, for this review, I took the VST version for a spin within Cubase Pro 12.

Inserted as an instrument on a suitable MIDI track, it will take extended MIDI notes as input (these serve as the base pitch for any subsequent randomisation) and then allows you to generate note sequences from this source by randomising note lengths, rhythms, pitch, velocity and pitch-bend, all with user control over the degree and style of the randomisation. This new MIDI data can then be routed to the MIDI input of a further MIDI/virtual instrument track for triggering playback or to be recorded. This latter option allows you to cherry-pick the best bits of Rhythmizer's output for further work.

In Use

Space precludes an all-encompassing walkthrough of all the options provided by Rhythmizer, but a simple example or two can illustrate some of the core possibilities. For example, lets imagine you wanted to generate some possible bass line ideas. Via the Rhythm tab, you might start with a fixed 16th-note rhythm and set the three Groove probability options to zero (these can be used later to dial in random rhythmic variations at eighth, quarter and half notes respectively). You can tweak the rhythm by 'dropping' notes using a (note) Density setting below 100 or by adding a positive Shuffle setting. Once you have defined the limits of the rhythmic variations, you can then switch to the Melody tab and dial in some pitch variations by picking a scale (there are lots to choose from) and adjusting the Prob setting. As you increase this, you are gradually increasing the probability that any note generated has a pitch other than that of the input MIDI note. The Steps control then limits the maximum number of scale steps away from the input note allowed. For bass lines, starting with a low value here (perhaps up to five steps?) works well, while for a top-line instrument, larger values allow more melodic movement. Then, once you have both rhythmic and pitch randomisation options set, you can start adding in Gate (so notes don't fill a complete time step) and velocity control, both with randomisation. And, if you want to experiment with some chord generation, the new Poly Stack feature provides for that possibility also with some interesting options for defining your own chord voicings. Finally, you can configure the Loop option, which allows you to essentially capture a short sequence of Rhythmizer's note randomisation process and then loop it for as long as the Loop button is engaged.

The second obvious application of Rhythmizer's ability to spit out user-controlled 'random' MIDI notes is to trigger slices created from a drum loop and that are mapped across a drum sampler instrument. I tried this using Cubase's Sampler Track (which has a neat slice mode) but it would also work with a sliced loop in Groove Agent or a similar drum sampler. It works well, although the settings required in order to generate something that is musically useful perhaps do require a little more thought.

Rhythmizer Dancer

Once you have got a basic understanding of how the key controls interact, there is a lot of fun to be had with Rhythmizer Ultra. OK, so you might not want to generate a whole album's worth of melodic, bass and drum part ideas based upon Rhythmizer's generative assistance but, given the very modest asking price, it's a cool and somewhat quirky tool to have available for an occasional musical nudge. If generative musical fun is your thing, then Futurephonic have a trial version of Rhythmizer Ultra available for download on their website.

summary

Rhythmizer Ultra is a cool little plug-in capable of generating all sorts of MIDI-based musical inspiration.

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Lindell Audio LiN2A

Valve Optical Compressor

Does Lindell's LiN2A sound as classy as the famous compressor that inspired it?

MATT HOUGHTON

I shouldn't need to state that Lindell Audio's LiN2A is based on the Teletronix LA-2A! A valve-amplified optical compressor, the LA-2A is a characterful and forgiving processor with a reputation for assertive level control that won't ruin the dynamics of a performance. Its character is due in part to the nature of the optical compression and in part to the sound of its valve stages. Several replicas are available, but the LiN2A, is priced competitively and it's pretty well specified too, with Cinemag input and output transformers, a Black Lion Audio T4BLA opto cell and, generally, 'name' components used throughout. As supplied, some of the valves vary from the models used in its muse, but internal jumper switches allow those who crave authenticity to install the original types.

Quick Tour

All the LiN2A's electronics sit inside a nice, solid 2U chassis and, as with most valve devices, it can generate quite a bit of heat: you'll want to leave space in a rack for this to dissipate through the top-panel ventilation slots. A tasteful two-tone front panel is largely dark blue-grey, punctuated by pale blue-grey legends, but that scheme is inverted in a middle stripe that hosts the company's logo and a generous, warm-glowing moving-coil meter.

This panel sports most of the expected controls, recreating the familiar LA-2A 'back to front' arrangement with a Peak Reduction knob (in effect, threshold, though it also affects the ratio a bit) on the right and a Gain (output level) knob on the left. Both are detented pots. Top-right is a rotary switch which sets the meter to indicate output level (referenced to either -10dBV or +4dBu) or gain reduction. On the left, a chunky toggle switch selects Limit or Compress mode. As on other such devices, these nominally give you an infinity:1 or 3:1 ratio, but the ratio, attack and release times all vary with the level and dynamics of the source.

I was surprised to see no Emphasis control. I've found this useful in refining an LA-2A's response to sounds with lots of high-mid and high frequencies such as esses and cymbals. Also, I'd half expected to find some other commonly added facilities, such as a wet/dry mix control, but there are no such embellishments.

On the rear, the main I/O (on balanced TRS sockets) are accompanied by a meter calibration pot, a grounding point and an IEC14 mains power inlet with 115/230V slide switch. Lindell deserve brownie points both for choosing an internal PSU, based around a toroidal transfomer, and for the inclusion of a front-panel toggle to switch the unit on and off — when the LiN2A's not in use you can power up your rack without having it heat the room!

Smoothly Does It

To test the LiN2A I tried both recording through it and inserting it in a DAW project or two, where I ran different sources through it. It's a positive sign that I have very little to say about the LiN2A! It delivered precisely the classy yet characterful compression I'd expect of an LA-2A. It's a wonderfully forgiving sound, and you really do have to work pretty hard to make it sound bad. This, along with the simplicity of operation, makes tracking through it a joy. As I'd expected, it was particularly effective on vocals, whether male or female, and in pretty much whatever register or genre I threw at it. But I also enjoyed it on both bass and acoustic guitars, and in that latter role the large meter came in very handy, helping me tame overexuberant strums judiciously. On rock drums I loved using it in the Limit setting as a sort of parallel 'crush' processor. (I tend to do parallel compression on separate mixer channels, though some might pine for a wet/dry blend control for that.)

Comparing the LiN2A with the equivalent UAD plug-in, the sound was very close indeed, though the hardware sounded that bit smoother when used at more extreme settings, for example for that drum crush role. As is so often the case, I also found it much quicker and easier to dial in a desirable sound using the hardware, though that may be a personal preference thing. I missed the emphasis control only rarely, so I don't think that's a big loss.

On the whole, then, this is a lovely, classy implementation of a studio classic. If you're in the market for a versatile compressor for your main recording chain, or want to get more hands-on when mixing, the LiN2A deserves consideration. ■■■

summary

A well built, classy-sounding clone of the LA-2A, all this lacks is the Emphasis control.

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Heritage Audio LANG PEQ-2

Inductor EQ

Although less well known today than the Pultec EQP-1A, the versatility of LANG's valve-less PEQ-2 won it plenty of admirers.

Heritage Audio
LANG PEQ-2
£859

PROS
- Offers greater flexibility and focus than similar vintage EQs.
- Superb at adding weight to low-end sources.
- Overlapping 'boost' and 'droop' options can be very creative.
- Good value.

CONS
- None!

SUMMARY
A great-sounding and surprisingly flexible vintage-style inductor EQ, for an attractive price.

NEIL ROGERS

After making their name with a range of high-quality Neve-inspired products, the last few years have seen Madrid-based Heritage Audio introduce more unusual vintage recreations, offering engineers the chance to get their hands on legendary recording tools at less than 'collector' prices. I reviewed a pair of their Motorcity inductor-based EQs last year, for example, and I now use those on an almost daily basis in my studio for recording and mixing. I didn't need too much encouragement, then, to take a look at their new LANG PEQ-2.

Heritage Audio have acquired the full rights to the LANG Electronics name and designs, and the PEQ-2 reviewed here is certainly the most well-known LANG product (though I'll be interested to see what, if anything, Heritage choose to bring back to life next!). LANG Electronics were based in New York in the 1950s and '60s and were a popular producer of after-market audio components for the broadcast and recording industries. In particular, they were known for supplying cost-effective parts for Ampex tape machines, which were expensive to maintain and repair. LANG soon began to produce their own products too, such as small broadcast-style mixers and equalisers, including the PEQ-2 'program equaliser'. Broadly speaking, the original PEQ-2 can be thought of as a non-valve, more cost-effective alternative to the more widely known Pultec EQP-1A inductor-based EQ. Both units were based around the same Western Electric design for a passive, inductor-based EQ, so its resemblance to the Pultec EQP-1A is perhaps unsurprising. As well being significantly less expensive than the Pultec, the PEQ-2 offered greater flexibility, with more frequency options available, and many engineers came to value it.

Features

Despite obviously being brand spanking new, the new PEQ-2 does gives off the air of an authentic vintage unit. On the front, it looks superficially the same as the original with the exception of the omission of the rack handles on the front of the vintage units. Heritage explained to me that they stayed faithful to the original circuity with the exception that they used their own custom input and output transformers and a small, external power supply. These modern, practical changes explain why this recreation doesn't feature the huge, heavily insulated external

input, output and power transformers that protrude from the rear of the vintage units. Designer Peter Rodriguez explained that they learned from their work recreating the Motown EQ units, for which they had to solve the issue of power transformers radiating noise into the passive components of vintage-style EQs. Implementing these adaptations allowed Heritage both to stay faithful to the sonic character of the original design and to offer their PEQ-2 for a relatively affordable price. The only other change from the original is the very welcome addition of 'soft step' controls.

As mentioned above, the PEQ-2 borrowed from the same Western Electric inductor-based EQ design as the Pultec EQP-1A, and in many respects they're similar tools. Controls-wise, the EQP-2 is a simple creature, featuring four EQ knobs that allow you to 'boost' or 'droop' low or high frequencies. There's also a bandwidth control that allows you to make the high-frequency boost more 'sharp' or more 'broad'. Many engineers will be familiar, whether in hardware or plug-in form, with this simple but effective style of EQ, with which the somewhat counter-intuitive technique of simultaneously cutting and boosting at similar frequencies can deliver surprisingly good results.

Compared with a hardware Pultec, however, the LANG has nearly double the amount of fixed frequency points to choose from (26 compared with 14), and so allows considerably greater flexibility in shaping the low end, as well as adding a 20kHz option for the high-frequency boost.

In Use

After patching in the pair of PEQ-2s that arrived at my studio for evaluation, their first outing was during a two-day tracking session that mostly involved recording a selection of acoustic and electric guitars. We all know you should endeavour to get the tone right at source when recording guitars, but having access to an EQ that can naturally change the shape and feel of what's coming through the speakers is still very handy when tracking. I found adding a little 5kHz combined with a little 'droop' at 2.5kHz seemed to work wonders on overdriven, picked guitar parts and, generally, I found that playing with options on the PEQ-2 was satisfyingly quick and creative when searching for the finer details in a guitar sound.

The EQ's next outing was on drums, and I was keen to see what the units had to offer on close-miked kick and snare, which is what I often use my Motorcity EQs for. You can add some serious low-end weight with a good inductor-based EQ and the PEQ-2 didn't disappoint. On my kick drum, I found boosting at 60 or 80 Hz, combined with a similar sized 'droop' at 100Hz, added an impressive sense of weight, and boosting around 120Hz while cutting at 200Hz on my 'top snare' mic added just the right amount of extra body.

All bass parts seemed to benefit from going through the PEQ-2, which is no surprise, but I was surprised with how often I found myself boosting more subsonic frequencies (around 30-40 Hz) combined with a 'droop' around 100Hz. This is often what people love so much about this style of EQ: you can get pretty heavy handed with bass boosts whilst still keeping your bass instrument or kick drum sounding tight and controlled.

In the more sedate mix setting, I found that I wanted to use the PEQ-2s in more subtle ways. I liked adding clarity with the 5kHz boost but often found that the settings lower down, for example 3.75 and 2.5 kHz, could be a bit intrusive if used more than subtly. Further up the spectrum, the 20Hz setting does a very nice 'air lift' thing, and I often found myself cranking the dial as far as it would go before it sounded too much — a great effect on a female vocal, and also for adding a percussive sheen to a strummed acoustic guitar.

I took being sent a pair of EQs for review as a hint that I should try them on the mix bus, so I mixed two projects through the LANGs during the review period. It was on the low end that I really appreciated them. Starting a mix with a low-end boost at 60Hz (regular readers will know I often use a 'top-down EQ' approach to mixing), I found myself needing to do less to individual sources in the mix. Introducing a 'droop' in the 100-200 Hz area as the mix became more developed allowed me to fine-tune where the low-end energy was focused.

At the other end of the spectrum, adding a high-end boost upwards of 10kHz worked well for an overall 'smiley face' mix-bus EQ. For the projects I was working on at the time, I often found that I had to be very careful when applying much high-end boost with the PEQ-2s so as not to pick up too much unpleasantness from cymbals and vocal sibilance. But given my success with manipulating individual sources, I suspect that with a little more time to experiment, I'd probably find a pair of these EQs to be more flexible in use across whole mixes.

Summing Up

I enjoyed my time with the LANG PEQ-2's a great deal. I've never been overly excited about using Pultec-style EQs because I often find them to be a bit of a one-trick pony (albeit a very good trick!), and what I liked most about the PEQ-2s was that their greater versatility encouraged me to use EQ in a more creative way. The LANGs don't quite have the same silky, effortless feel that a Pultec can have when boosting the top end, but I far preferred the PEQ-2 on the low end — it allows you to add often dramatic amount of weight in a very focused way. These EQs worked superbly for me as a studio 'all rounder' and sit in that nice place of being a creative, simple-to-use vintage-style EQ, whilst offering enough versatility to be useful on anything coming through your monitors. The price is also attractive — this is a full-fat, standalone EQ but it's priced comparably to many 500-series modules. The LANG PEQ-2 is well worth further investigation. ■■■

Replacing the vintage power supply with a more reliable modern one enabled Heritage to fit all the components inside the 2U chassis — leaving very little of interest on the rear panel!

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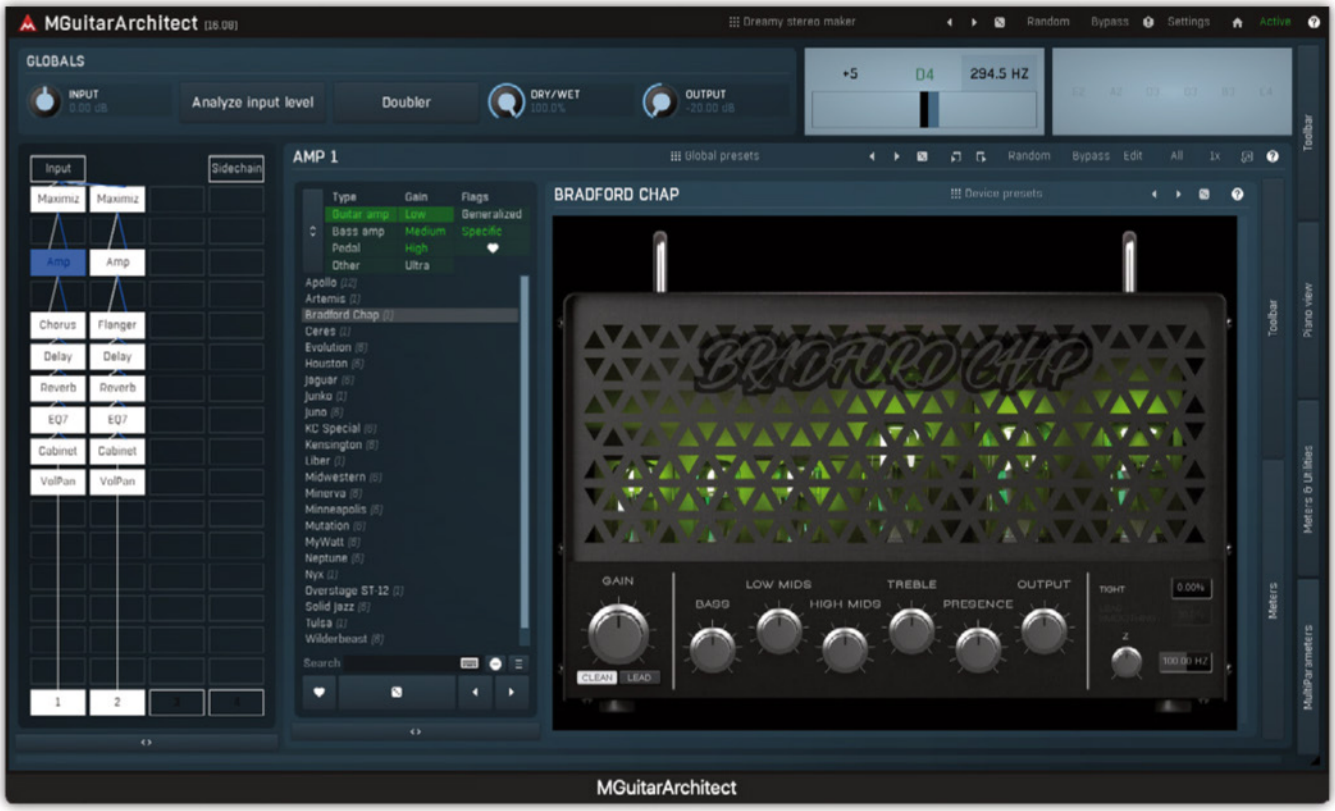

Melda MGuitarArchitect

PAUL WHITE

Amp & Effects Modelling Plug-in

With up to four parallel chains available, you're free to design pretty much any amp and effects chain imaginable.

Melda's MGuitarArchitect is available separately or as part of the company's various bundles, and includes all the company's existing MTurboAmp and MCabinet devices (those who already have these plug-ins can upgrade at a discounted price). As with other Melda plug-ins, it's available for both macOS and Windows, and supports AAX, AU and VST (2 and 3) plug-in hosts. As well as boasting impressively flexible routing options, it comes with a large selection of amps, miked cabinets, studio processors and effects pedal emulations.

Of course, the world isn't short of guitar amp emulations, so what makes this one worth investigating? Well, what's most different about this one is its Chain section: the input signal can feed up to four 16-slot signal chains, which are arranged as four vertical columns of cells, and the fourth of these can be used for side-chaining. Clicking on an unoccupied space gives you the option to populate it with an amp and/or overdrive plus stompbox effects, studio effects or utility effects, all of which are organised by category. You could, in theory, run anything up to 64 effects in various series/parallel combinations across the four chains.

Overview

There are dozens of amp and cabinet models to choose from, and a very generous selection of stompbox simulations too, each equipped with a sensible set of controls that allows plenty of scope for tweaking without getting lost down the rabbit hole. While some of the hardware amp inspirations (such as the MCM800) are easy to guess at, I found most of the other names somewhat cryptic, but once you figure things out you'll find that all the main bases are covered, from super-clean to uber-filth. Each module comes with its own presets that make patch creation pretty easy, and user presets can be saved for the whole effect or for each chain individually. Furthermore, a selection of other Melda Productions effects plug-ins that you already own (eg. if you get this as part of a bundle) can be called up and deployed inside the plug-in, which is a nice touch.

One of the modules included in MGuitarArchitect is MConvolutionEZ, »

Melda MGuitarArchitect
£286

PROS
- Good-sounding amps and cabs.
- A dynamic playing feel.
- Lots of creative routing opportunities.
- Free addition if you own a qualifying Melda bundle.

CONS
- Some amp names could be more informative.
- Drag-and-drop routing would be nice.

SUMMARY
Authentic sounding amps, cabs and effects with plenty of routing flexibility. There's a lot to like here.

evo 16

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AUDIENT

» a treatment based on impulse responses or IRs to add spatial and microphone characteristics based on a range of classic models. Should you feel the need for even more scope, the third-party Neural Amp Modeller (available as a free amp profiler plug-in from Steven Atkinson) may also be used to load new amp models; there are three Neural amp models available directly from within MGuitarArchitect.

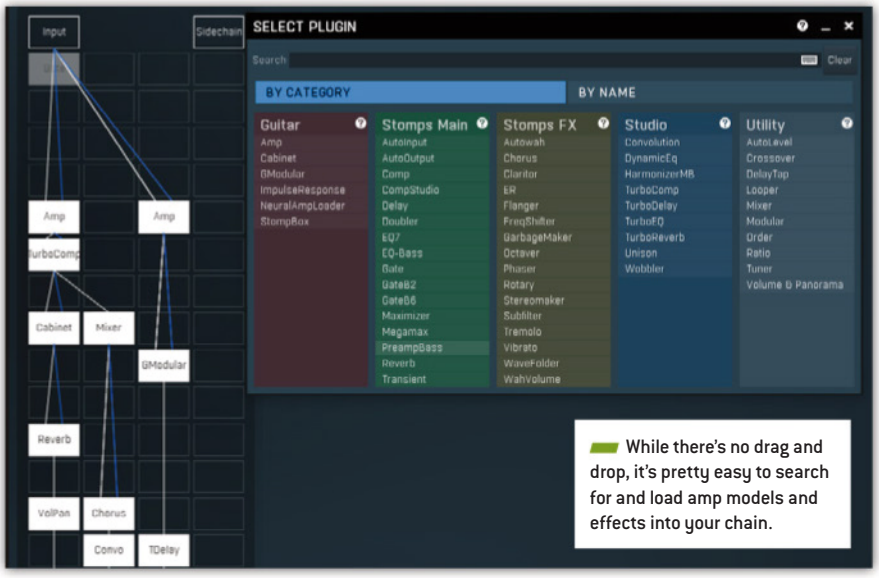
When you call up an amp or pedal, or select one already deployed by clicking on it, you see a photo-realistic control panel with presets listed on the left. The utility modules such as Vol/Pan show a more simplified interface. Some modules, such as those using convolution, have the option to open up additional advanced settings. You also get an accurate tuner with a polyphonic readout and an auto input gain setting. Global settings include various options for conserving CPU overhead or using oversampling.

By default, adding a module results in it taking its input from the previous module in the same column. However, right-clicking on a block opens its settings window, in which you can set up various parameters such as side-chaining and feedback, and change the default routing to pick up its input from a block above in another row. For example, you could have an amp emulation in row one and split its output to feed four different cabinets or four different effects chains. It's perhaps not as intuitive as being able to drag patch cables, as some plug-ins allow, but it works well enough.

As you have so many modules from which to choose, figuring out what combination gives you the desired end result can take a bit of time, but the excellent range of presets provide great 'jumping off' points. There are all the tools needed to create metal guitar, classic rock and blues sounds as well as contemporary pop, but thanks to the flexible routing and generous library of stompbox emulations, those into more experimental sound design won't be disappointed. You can even create a modular effects matrix and then drop it into one of the slots, as though it were a single effect. Overdrive-type stomp effects tend to be listed along with the amplifier types; speaking of which, you can browse by Low, Medium, High and Ultra gain amplifier types. A lot of the amplifiers seem to tend towards high-gain sounds but there's still plenty of choice if you want something more restrained.

The included speaker cabinet emulation is based on existing Melda Productions technology and, in conjunction with the responsive amp models, it's capable of creating both the right sound and the right playing feel. Unlike straightforward IRs, there's a lot of scope for adjustment here, and this has a significant impact on the overall tonality. When it comes to effects, the large library of stomp pedal emulations cover all the familiar ground. There's not much in the way of really oddball effects, but you do get a broad selection, including a neat rotary speaker effect, a frequency shifter and a pitch-shifter/harmoniser. Of course, the ability to create parallel effect chains means there's still plenty of scope for creating complex effects of your own — and to help you in your multi-effects quest, there are various utility modules such as Crossovers, a Looper and so on.

As well as the built-in effects, MGuitarArchitect allows you to load, internally, any other Melda plug-ins you own — pictured here is MTurboReverb.



While there's no drag and drop, it's pretty easy to search for and load amp models and effects into your chain.

Verdict

MGuitarArchitect is nothing if not comprehensive and, importantly, the combination of amp and cabinet models delivers a touch-responsive playing feel combined with the ability to cover pretty much any guitar tone you might wish to recreate. Those looking for something beyond the conventional can also look forward to long and rewarding adventures when exploring the potential of the chains and their routing possibilities.

£ £260 (discounted to £86 when going to press). Also included in Melda's MCreativeFX, MTotalFX and MComplete bundles.
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How I Got THAT SOUND

Howard Jones ▪ 'What Is Love?'

JOE MATERA

Howard Jones burst onto the scene in the early 1980s with a collection of era-defining synth-pop hits such as 'What Is Love', 'No One Is To Blame' and 'Things Can Only Get Better'. Asked to select a favourite sound, he circles back to the first of these.

"It starts off with a Yamaha DX7 bass sound — a synthesizer that had just come out at the time. I think I'm right in saying that 'What Is Love?' was the first hit song that had featured the DX7. I know that other people had used it on 'B' sides and things like that, but 'What Is Love?' was the first big hit song that actually featured the DX7.

"The bass sound is such a great, solid bass sound, courtesy of the the DX7. But the DX7 was impossible to program, so I'm not going to say that I used anything except the preset, which is the cardinal sin among synth players. But I didn't have time to do 10 years' research to be able to program it. There's also a brass sound that I used on the DX7 as well, that 'da da da da da da da da da' melody. And then there is the arpeggio background figure, which was done on a Roland Juno-60.

"The drums were done on an E-mu Drumulator, and we kept the tom sound from that, and then [*engineer*] Steve Tayler and Rupert [*Hine, producer*] had an AMS digital delay, and they would trigger a more beefy, substantial snare sound and bass drum sound in the AMS unit. I think they had 16 seconds of sampling back then, as this was the early days of sampling. Then there's the shakahachi sound, which was done on an Emulator II. The initial brass fanfare pads were a combination of a Juno-60 and a Roland Jupiter-8, which obviously were my bespoke sounds for all those things."

Tales Of The Unexpected

"There's a little 12-note melodic sequence that was done on an SCI Pro One. When we get to the mid section, it is really unexpected, as the song goes totally off-piste, into a really weird area, which is one of my favourite parts of the song. I did that on the Moog Prodigy. They cost 250 quid at the time and I had two of them, one for bass and one for leads. There is also an amazing synth sound, a raucous, crazy synth sound in the middle of that section that I got by using the mod wheel when I paired the two oscillators.

"The hi-hat part you hear wasn't programmed. That was Trevor Morais, who owned Farmyard Studios, and put together the two-bar hi-hat pattern by sampling it in the AMS. He gave it a bit of a 16th-note swing, which gave the whole song a really cool groove. It was just a two-bar loop that repeated, being retriggered all the time. But it's quite a significant part of the groove of the song. The big vocal sections were multitracked, which we did via a slave reel in order to multitrack all the vocals. And Rupert sang with me too, and it gave another texture to the backing vocals.

"At the Farmyard, they had one of the first SSLs, and 'What Is Love?' — and *Human's Lib* — was the first thing that Rupert and Steve had actually recorded and mixed on an SSL. They used the built-in compressor on the SSL, to really give the sounds an extra punch. When I listen back to that record now, and when you hear it on the radio, it just absolutely kicks out. It really is a fantastic-sounding record and a tribute to Steven and Rupert."

Hear The Sounds

W https://open.spotify.com/track/5YBbCAsC19yTItMOaRImG1
W www.youtube.com/watch?v=G9KZg-mFPgI

Photo: Simon Fowler

12

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Audio-Technica 3000 Series Wireless In-ear Monitor System

Licence-free wireless is fine for smaller shows, but more serious call for a more serious system.

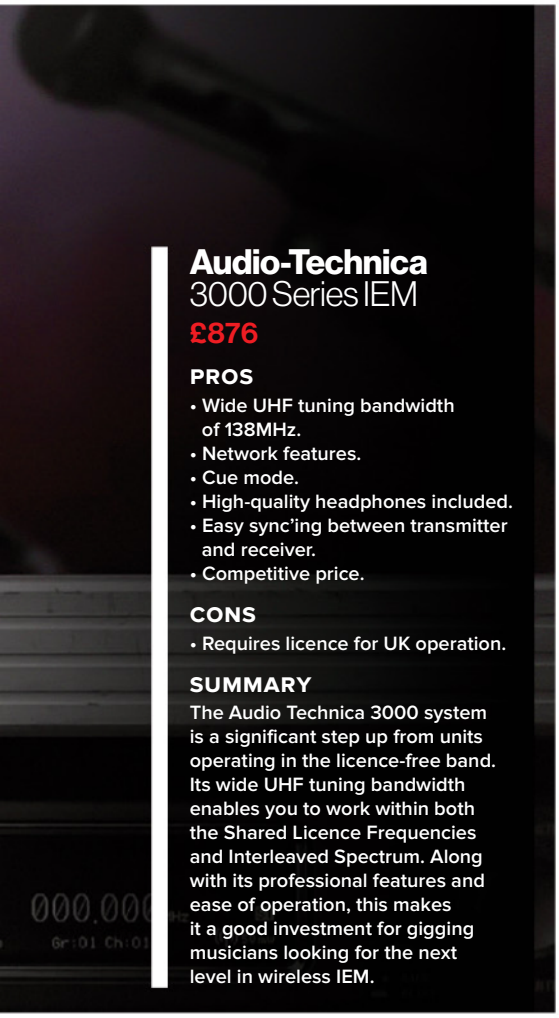
MARK GORDON

From their humble beginnings in the 1960s manufacturing turntable cartridges, Audio-Technica have grown into a significant force in the audio industry. Though the company's feet are still firmly rooted in the turntable market, over the decades they've become a leading name in pro audio and their catalogue has expanded to include a wide range of headphones and microphones, in both consumer and professional audio markets.

Wireless technology has also played a big part in Audio-Technica's story — which brings us to the subject of this review, the new 3000 Series Wireless In-ear Monitor System. With elements based on the 3000 Series UHF Wireless Microphone System, Audio-Technica say it has been designed to bring "professional sound quality and features to all levels of performers and performance venues".

Whys & Wherefores

If you're currently using one of the many wireless IEM systems that make use of the 'licence-free' UHF frequencies available to us in the UK, you may well have experienced some level of interference, especially if you're gigging with your own band or bands and you're all using IEMs or radio mics. Without going into too much technical detail, the licence-free frequency band is very narrow (863-865 MHz) and can only accommodate a limited number of devices — around four — before harmonics generated by one device start to interfere with another, resulting in distortion, artefacts and reduced range. The solution is to use the areas of the UHF band that are less congested. However, this does come at a cost, both in terms of upgraded hardware and of licence fees (the latter not as onerous as you might expect — see 'Finding Frequencies' box).



Audio-Technica 3000 Series IEM

£876

PROS

- Wide UHF tuning bandwidth of 138MHz.
- Network features.
- Cue mode.
- High-quality headphones included.
- Easy sync'ing between transmitter and receiver.
- Competitive price.

CONS

- Requires licence for UK operation.

SUMMARY

The Audio Technica 3000 system is a significant step up from units operating in the licence-free band. Its wide UHF tuning bandwidth enables you to work within both the Shared Licence Frequencies and Interleaved Spectrum. Along with its professional features and ease of operation, this makes it a good investment for gigging musicians looking for the next level in wireless IEM.

The Audio-Technica 3000 system is one that does operate in the 'licensed' bands and offers exceptionally wide-band UHF tuning bandwidth of 138MHz across one of two frequency bands, DF2 (470.125-607.875 MHz) and EG2 (580.000-713.850 MHz). This enables the use of frequencies that should ensure good, interference-free operation.

In addition to its wide UHF coverage, the 3000 Series also includes features such as networking, wireless synchronisation between transmitter and receiver via infra-red, and a Cue mode (see 'Join The Cue' box) that allows audio engineers to monitor multiple channels with a single receiver. These take it considerably beyond what you'll get from a typical licence-free system.

Send & Receive

Arriving in a large cardboard box, the 3000 Series ships as a complete 'ready to go system' comprising the ATW-T3205 stereo transmitter, the ATW-R3250 stereo receiver pack, and the ATH-E40 in-ear headphones. The transmitter can be rackmounted and ships with the appropriate 19-inch rack ear extensions, but straight out of the box it is a free-standing unit measuring around eight inches square and powered from an included 12V wall-wart PSU.

The rear panel hosts left and right input XLR sockets, alongside left and right Loop Output quarter-inch jack sockets, which mirror the input signal. These outputs can be connected to speakers or a recording device but, as their name suggests, they can also be used to configure more complex monitoring setups by routing the signal to a second transmitter.

Accompanying the familiar bayonet RF socket, to which the supplied flexible UHF antenna is connected, is a network port. This allows the transmitter to integrate with other Audio-Technica network-enabled devices and make use of Audio-Technica's Wireless Manager software. This comprehensive Mac- and Windows-compatible application enables you to create a detailed frequency plan by scanning the RF environment and deploying an optimal set of frequencies to each connected device.

The front panel is minimalist, with only three buttons, including the power switch. The panel is dominated by a large OLED screen that displays a trove of information, including the transmitting frequency, group and channel, audio signal levels and RF status. You can also name the transmitter (Guitar, Vocals, etc) and have that displayed as the dominant element on screen.

A large data-entry wheel with push confirmation is used to scroll through all the menus and adjust parameters. A small button to the left acts as a 'back' button and doubles as an 'RF off' button with a long push. The final thing to note on the front panel is a small infra-red sync window to the left of the OLED display. This enables automatic frequency synchronisation with the receiver pack.

The receiver body pack feels solid and well made, to a design that will be familiar to users of IEM systems, featuring an antenna, a 3.5mm headphone jack and an on/off volume knob across the top. Interestingly, the headphone cable can act as a second antenna, which, although not providing 'true diversity' (which requires a second receiver), does offer the possibility of a better and cleaner signal, with the system most of the time using the main antenna but occasionally switching to the audio cable as backup to avoid a drop-out.

A small function button turns on the integrated OLED screen (which shuts off automatically after 30 seconds of inactivity). With a second push, the same button brings up the audio setup screen. Pulling down the two catches either side of the pack opens the hinged battery cover, to reveal space for two AA batteries. Below that is a control panel featuring a dedicated Sync button alongside Up, Down, Back and Set buttons, for navigating through the menus. There you can adjust settings including the receiver's frequency and mono or stereo operation, as well as saving and recalling presets. The Audio menu gives access to high and low EQ, which can be set to 80/160/320 Hz and 6/8/10 kHz (±9dB) and a limiter that can be set between -30dB and off in 6dB steps.

On the rear of the receiver is another infra-red sync window, which needs to be held facing the corresponding window on the transmitter to sync the two units. There is also a charging terminal compatible with the ATW-CHG3 two-bay charging station. If you choose to use NiMH batteries, this allows two receiver packs to be charged together, and will automatically turn off if alkaline or damaged batteries are detected.

Ear Test

Interestingly, Audio-Technica have chosen to include their entry-level in ear monitors, the ATH-E40s, as part of the package, perhaps because they anticipate that most users will have their own custom moulds or personal choice of headphones. Having said that, the ATH-E40s are certainly of high enough quality for the professional musician.

»

Join The Cue

One of the receiver's menu options is Cue Mode, a great feature that enables monitor engineers to listen to several different IEM mixes from multiple transmitters through one receiver pack. Ten presets can be stored in the receiver pack, each comprising frequency, name, group and channel, RF mode, AF mode, and balance. When you select Cue Mode on the receiver, a small preset number appears in the top-left corner. Pressing the Function button on top of the receiver pack scrolls through each saved preset, enabling you to audition each transmitter's mix.

Finding Frequencies

The A-T 3000 Series uses the UHF band of frequencies to transmit and receive audio. Originally used for analogue TV transmission and separated into 8MHz chunks that are still referred to as channels, the usable UHF band spans a range from 470-865 MHz. From these frequencies, only a select few are legal for public use in the UK, so it is crucial to know on what frequency you are operating your radio microphones or IEMs.

In the UK, the UHF band is regulated by Ofcom, and they dictate which channels are free to use and which require a licence. Fines can extend up to £5000 and you can have your equipment seized if you're in breach, so if you're working with wireless UHF it's important to know if and when you need a licence. It's also important to know that UHF licensing rules differ between countries, so, for example, channels that are free in the USA may require a licence in the UK.

Let's start with the licence-exempt UHF frequencies that you can use for free within the UK. These occupy channel 70 and span 863-865 MHz. As you can see, this is not a particularly wide range, so it has a limit of around four wireless systems before you start to experience interference.

The next step up is the Shared Licence Frequencies, which occupy channel 38 of the UHF band (606-614 MHz) and are dedicated to wireless mics and IEMs. Using frequencies within this range requires a generic licence costing around £75 a year, but the extended range allows for 10 or 12 systems to work comfortably without inference. Bear in mind, however, that this is still a 'shared space', so if there are other users nearby there could be a clash of frequencies. Finally, we have the Coordinated Frequencies, which span from 470-790 MHz, known as the Interleaved Spectrum. This uses the spaces available between the digital TV channels (and other potential users, such as the military) with frequencies allocated by Ofcom on a per-location and per-date basis. This means that you can be sure the frequencies you're using in this range will be free from outside interference and not being used by anyone else. Large touring companies and theatre productions that need 50 or 60 channels of wireless would typically use this option. Costs vary, depending on whether indoor or outdoor use is required, but typically you'd be looking at £50 for a block of six frequencies over 48 hours. The wide bandwidth offered by the Audio-Technica 3000 system has an advantage here, as it covers such a big range of frequencies.

» Packaged in their own zipped hard case, they feel solid and well made, and they look very classy. A significant part of the casing is semi-transparent, which gives a good view of the internal workings and twin-driver, push-pull design, where the two drivers are mounted facing each other, resulting in increased efficiency, faster attack and a more linear, controlled response. The cables are detachable from the earpieces and have the usual thicker section that fits over the ear. This is more malleable than I am used to, allowing plenty of adjustment and, consequently, a comfortable and secure fit.

Included are three sets of silicone ear-tips in various sizes. At the risk of sounding like a broken record, and having reviewed several IEMs recently, achieving a good fit is vital in ensuring uncompromised performance with any IEM. I found the mid-sized tips worked best for me, and I was able to achieve a good level of isolation and comfort. In previous IEM reviews I have used foam tips, which naturally expand in the ear and offer something closer to the custom-moulded experience. While the supplied silicone tips certainly couldn't be described as bad, from a personal point of view I would consider experimenting with foam tips, as this may enhance comfort and fit.

If you'd like a little more information, the ATH-E40s (along with the ATH-E50s and ATH-E70s) were covered in some detail by Mike Crofts in 2016: www.soundonsound.com/reviews/audio-technica-iem-series.

Up & Running

There are a number of different ways to configure the 3000 system, depending on your requirements. Initial setup couldn't be easier, as both the transmitter and receiver are set to the same frequency at the factory, so it theoretically works 'right out of the box'. As explained in the 'Finding Frequencies' box, though, it isn't just a case of picking a frequency and away you go! However, the 3000 system includes several features that make this process very easy.

If you're using one system and know the frequency you'll be working with, you can simply dial this into the transmitter manually, using the data-entry knob, and set the receiver to the same frequency. For multiple IEM setups, you can perform an RF scan to analyse the surrounding frequencies, which will let you know how many channels are available in each of the 18 Groups — a Group being a collection up to 25 channels (frequencies) that can work together without interference caused by intermodulation.

An RF scan is initiated from the Scan menu on the receiver pack, and took around a minute in my case. Once it's completed, the receiver pack prompts you to press the Sync button and transfer this information to the transmitter, by aligning the two IR windows on each device. The transmitter then displays a list of the available Groups and the number of open channels (frequencies that are free from interference) in each group. So, if you're configuring a setup with a five-piece band all using IEMs, you would want to use five free channels all within the same Group. You can also view the RF scan as a graph on the transmitter screen, which gives a more detailed view of the whole RF environment.

A long press on the transmitter's rotary controller brings up the RX Sync window, which enables synchronising of the receiver to the selected Group and channel. The RX Sync window not only transmits the frequency information to the receiver pack but also gives the option to change several other settings, including EQ, limiter settings and stereo/mono operation. If you have multiple transmitters and receivers connected via the network port, you can use the Deploy option to set the frequency of all transmitters in the same network at once.

With both receiver and transmitter set to the same frequency, the small LED on the top of the receiver pack turns green and the RF level on the display (which looks and operates much like the signal-strength display on your phone) will show a signal.

In Use

I connected the ATH-E40s directly to a headphone amp, to try them out prior to using them with the wireless system, and found they offered a clear and well-balanced sound. The highs had plenty of top-end detail, midrange frequencies were pronounced but not harsh, and the bass was very much present but not overpowering. The ATH-E40s don't overwhelm, but neither do they have you pulling them from your ears in disgust. This sounds like damning with faint praise, but actually it's exactly what you want (and expect) from a high-quality entry-level IEM.

Used as part of the 3000 wireless system, the ATH-E40 offered, as far as I could tell, identical performance. The receiver introduced a very small amount of hiss at high volumes, but this was only perceptible with no audio playing.

Boosting the sensitivity on the transmitter by a few dB and lowering the receiver volume rendered the noise negligible and the ATH-E40 performed perfectly.

Different locations with different RF environments will have a bearing on how well the 3000 Series IEM system works, but during my limited test I was very impressed with the performance. With the unit set to the lower 10mW RF power, I was able to walk almost 100 yards away (with the signal passing through several walls) before I experienced the slightest dropout, with the audio signal remaining clean and consistent throughout. A higher RF power of 50wM is available, and would potentially offer an even greater range. However, keeping the RF power to a minimum helps reduce intermodulation distortion and means that more channels can be used cleanly.

Go Pro?

The Audio-Technica 3000 IEM system is a big step up from the licence-free, entry-level IEM systems many of us are used to using and, inevitably, costs more than such systems. However, if you're finding you're hitting the limits of the licence-free offer, the 3000 has a clutch of features that could well tempt you in its direction. These include an extremely wide frequency range, lots of options for interference-free operation, and the ability to use either Shared Licence Frequencies or Coordinated Frequencies (see 'Finding Frequencies' box).

"I couldn't fault the audio quality of the system at all, and the range extended far beyond what anyone would reasonably need."

The majority of comparable IEM systems (in terms of price and features) that I could find working outside of the licence-free frequencies appear to have a much narrower UHF range, typically focusing on the 606-614 MHz band. Although this would be enough for most semi-pro or smaller professional situations, the extended UHF range offered by the Audio Technica 3000 system can only be a good thing.

I couldn't fault the audio quality of the system at all, and the range extended far beyond what anyone would reasonably need. Setup of the 3000 system is also made very easy thanks to the RF scan and Sync features.

If you're a monitor engineer, the Cue Mode feature (see 'Join The Cue' box) is a fantastic way to keep on top of each IEM mix, and the integration with Audio-Technica's networked wireless environment takes the system yet further into the pro arena.

If you're looking for a high-quality and affordable IEM system that has the potential to fit into a more professional setting, the A-T 3000 Series really ticks the right boxes.

£ £876 including VAT.
T Audio-Technica +44 (0)1132 771441
W www.audio-technica.com

Kid Harpoon

Songwriter & Producer

The man behind colossal hits for Harry Styles, Miley Cyrus and Florence + the Machine takes a surprisingly old-school approach to writing and production.

PAUL TINGEN

"Our job is to make people feel something. You don't need to be the best musician to do that. You can make someone feel something with two or three notes played slowly, or a song with just two chords. Harry Styles' song 'Golden', which is one of my favourite songs that we have done together, has just two chords. I'm a professional, but I also try to maintain a degree of unprofessionalism. Otherwise, things get too shiny.

"You can go on TikTok or Instagram today, and there's always some guitarist or piano player who is insanely, mind-blowingly good. It kills me, and I think, 'Oh, I suck. Why can't I do that?' But it doesn't make me feel anything. It just makes me jealous that I can't do the same thing. However, the goal is not to be good. The goal is to make you feel something. It's important to remember this.

"Having said that, some people can be good in a way where they contribute to

the music being incredible, and making you feel something. The Michael Jackson stuff is incredible, because amazing musicians and jazz musicians are playing with a pop mentality, and everything crosses over into a world where you can appreciate the musicians from a technical point of view, and the songs are great and they make you feel something. To me, that's the pinnacle of music."

Tom Hull, aka Kid Harpoon, is one of the world's biggest songwriters and producers, as acknowledged by his 2023 Songwriter of the Year Brit Award and two Grammy Awards. He co-wrote and co-produced Miley Cyrus' megahit 'Flowers', and dominated the charts in 2022 thanks to his work with Florence + the Machine, Maggie Rodgers, Lizzo and, most famously, Harry Styles. Hull co-wrote all but one song and co-produced all songs on *Harry's House*, including the ubiquitous single 'As It Was', which spent 10 weeks atop the UK charts and 15 weeks atop the US charts, and was Spotify's most-streamed song in 2022.

The Ideas Man

Rather than rely on programming, samples and DAWs, Hull prefers to write songs with several people in a room playing traditional instruments, and using analogue gear to record them — again, with the emphasis on feel over technique. "The hardest thing in the studio is when the best guitarist in the world walks in, and everyone just accepts that they're going to play guitar. But that's not necessarily right. You don't necessarily want the fastest or the best, you want the correct idea and something that feels exciting. Someone could play three notes really badly, but put it through Ableton and pitch it around, or put it through a sequencer, and you're like 'Holy shit, that's amazing, that's the vibe.'

Tom Hull, aka Kid Harpoon, co-wrote all but one of the songs on Harry Styles' hit album *Harry's House*.

"It is helpful to have skills. When you have skills and knowledge, you can experiment with it. It gives you more tools to work with. When I was younger »

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» I was into punk, but my school was very classical with grades and theory, and you needed to be playing cello or clarinet or something. Guitars were nothing. It made me stick up my middle finger to music theory. But as I got older, I was like, 'I wish someone had shown me this.' Because now I enjoy it. You learn one new chord, and at some point you remember it and put it in a new song, and it's cool. Learning inspires the creative side.

"It is also helpful to have the right gear. I'm a very out-of-the-box producer, and I have a lot of outboard in my studio. I come from the original approach to recording and like to record with microphones. I play guitar, piano, drums and everything, and have live amps, and so on. I also do a lot of programming on MPCs, and I have a ton of drum machines and synths. And I use samples and try to make things sound modern. It's not like I want to be outside the box making pure live-sounding music, but I find that it's quicker and more exciting to play a synth, or to program a drum machine, or to play a drum beat, and to chop it up and edit it and place it where I want it, rather than use an Ableton station to come up with everything. There's something about a drum kit and a drummer and that mindset that makes a drum beat way more tied together. Otherwise you end up thinking 'kick drum' and then 'snare', 'hi-hat' and 'percussion', and then another snare is going to come in in the chorus. Even if you end up replacing all your drums, when sitting behind a kit you can feel a rhythm, and make it feel more natural."

Out Of The Blocks

Hull's philosophy has its roots in his early years in music. He was born in 1982 in Chatham in Kent, and started playing guitar when he was 10 years old. The instrument has remained central to his musical life.

"I started on acoustic, but also wanted to be like Jimi Hendrix or Stevie Ray Vaughan. I learned to sing and played some piano as well, and started performing live early. My first recording experiences were with Jim Riley at his Ranscombe Studios in nearby Rochester. He recorded to tape, and was very focused on the performances. Around that time I did a show under the name Kid Harpoon, after a character in a story that I had written, and the name stuck.

Kid Harpoon's only album to date was produced by Trevor Horn, an experience which made its mark on the young artist.

"What I did at the time was all acoustic, though I approached that very different differently than most. When someone goes on stage with an acoustic guitar your heart kind of drops because you expect it to be all sad and introverted. I wanted to do be the opposite of that. I was into Gogol Bordello and other indie punk bands, and wanted to have that kind of energy."

Kid Harpoon: "I'm a professional, but I also try to maintain a degree of unprofessionalism. Otherwise, things get too shiny."

When he reached his 20s, Hull moved to London and joined the music scene there, particularly at a live venue called Nambucca, where he became resident songwriter. He toured as a guitarist with acts like Jamie T and the Kooks, and in 2006 signed with the independent label Brikabrak, which released two EPs under the name Kid Harpoon. In 2009, Hull signed to the Young Turks label, which released his first solo album, again under the name Kid Harpoon. *Once* was produced by the legendary Trevor Horn.

"When it came to making my records, everything was fully Pro Tools, and I didn't know what to do. I tried all sorts of different things, and we actually made a version of my album on tape, and then transferred things to Pro Tools, but we messed it up because we'd recorded too quiet on tape, and when we then in Pro Tools boosted everything, there was a ton of noise. I realised that I didn't know how to do the studio thing. So I tried to find a producer.

"Trevor Horn's publishing company, ZTT, had signed my publishing. He explained to me what we had done wrong with the recordings, and said, 'Let's go to LA. I'm going to put a band together and we'll record it in just two weeks, because you already have the songs.' We recorded the album at his house in LA, and did some mixing at SARM in Notting Hill.

"The record came out great, but I felt during the making that I had missed my moment. As an artist you're looking for a wave and when you find one, and suddenly something goes viral, you surf that wave, and then the next one. When making that album I realised deep down, 'I missed my wave.' When you're working on a project, you need everything to be rushing forwards so that when it comes to the release day everything comes together. But I'd been touring without an album and my momentum was gone.

"I didn't know what else to do, but when watching Trevor I felt this spark inside. 'Oh, that's what a producer does. That's incredible.' He's one of the best producers I've ever seen in the studio. He's kind of brutal and demanding but amazing at the same time. Just the way he talked about stuff. 'This is the bit where the fireworks happen.' 'This is where we do this.' I was like 'Yeah, that's how you make records.' It finally clicked. I made the transition from artist to songwriting and producing for others at that point."

Standing On Ceremony

Hull's career journey took the detour of being a musician for hire, which in turn delivered a stroke of luck. "As part of the music scene in the UK I toured with Dizzee Rascal, Adele, Florence, and others. I'd worked with Florence on some lyrics and music before, and when my artist career wasn't going anywhere, I reached out to her and suggested we do some writing. Her A&R and manager were not too impressed, as I was unknown, but I got half a day with her. I was like, 'This is my break, I need to go for this,' and spent ages and ages working on a song we had started. »

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Florence + the Machine's hit album *Ceremonials* provided Kid Harpoon with his big break.

» "Her A&R thought the song had potential, and by this point Flo was working with Paul Epworth, so I went in with them to work on the song. Paul's very inspiring to be around when he's in the zone, and he brought his magic to the song, which became 'Never Let Me Go'. Then he was like, 'Do you want to come back tomorrow?' We made another song, 'Leave My Body', and some lyrics that I had written for 'Never Let Me Go' ended up getting used in 'Shake It Out'. So I ended up with three cuts on *Ceremonials*. Paul was like, 'Man you're really good at this, you should focus on this. Try not to do everything. Do one thing really well.'"

Ceremonials was a huge commercial and critical success, and 'Shake It Out' earned Hull an Ivor Novello nomination. Hull's work on the album opened many doors for him, and in 2013 he enjoyed songwriting and occasional co-production credits with Jessie Ware, Calvin Harris and many others. "'Wildest Moments', which I did with Jessie, brought me a lot of attention, and I'm still really proud of that song. It's the same with 'Sweet Nothing', which I did with Calvin and Florence. Calvin had sent a track, and Flo and I worked on it at Trevor [*Horn*]'s place and sent what we had done to Calvin. A few weeks later he had restructured it a little bit and created this whole new track underneath it, and it blew my mind. So I was having those hit songs, and then other things started to come in, like the songs with Shakira and Years & Years, and so on. Everything was growing."

Hull's career as a songwriter and producer has since taken in credits like Shawn Mendes (including 11 tracks on the singer's *Wonder* album from 2020), Lykke Li, Maggie Rogers and many more, including the aforementioned Miley Cyrus, Lizzo, Florence + the Machine and Harry Styles.

Beats Working

"A lot of my writing starts with a piano and acoustic guitar," says Tom Hull. "Nowadays you have producers who say, 'Hey, I've got this great beat,' and they pull something up and you write a melody on top. I like working like that as well. But I do like writing with an instrument. Harry and I wrote the song 'Falling' at the piano in 30 minutes, and we wrote 'Sweet Creature' with me on an acoustic guitar. I find that it helps if you've got a really good acoustic guitar or piano part, instead of just playing chords. There's a Harry Styles song called 'Matilda' that started with a great acoustic guitar part, and we wrote it around that.

"Writing on one instrument has that classic thing that if a song sounds good on just acoustic guitar or piano, it's great. The problem is that sometimes when you write like that, it can get hard to produce it. You try things, and then you're like, 'Ah. It just still sounds better just with a piano and a vocal.' I think that writing on an instrument as opposed to starting with a beat gives a bit more consistency to the song. When you're producing a track in a DAW, it tends to be split up in sections, like 'Here's the chorus, here's the verse.' But when you're writing on an instrument you're moving chords and melodies in a more seamless way, as opposed to block, block.

"But both have their merits, and there's something really exciting about a song being written and produced at the same time, because sounds encourage you. Also, you may not be sure a song is good, and the production can help you go, 'I can see it now, it's going to be great,' and then it pushes you to finish a lyric or a melody.

"Like 'Flowers', which Miley had written with Michael Pollak and Aldae as a piano ballad. Miley came to our studio, Tyler jumped on the synth and I jumped on bass, we got a rhythm going and I got the bass line, and she sang along. It had a cocktail vibe, but we also

Harry Styles' *Fine Line* contained the hit single 'Watermelon Sugar', co-written and co-produced by Kid Harpoon.

wanted it to sound modern, so we played and programmed drums. We took a great piano song and managed to turn it into a disco-type track."

Creature Comforts

Hull started working with Harry Styles in 2017, when the two co-wrote the song 'Sweet Creature' at Hull's then studio in LA, Harpoon's Barn. The production was completed by the trio who produced the whole of Styles' debut solo album: Jeff Bhasker, Alex Salibian and Tyler Johnson, with help from Hull on 'Sweet Creatures' and 'Carolina'.

"It was the first time I worked with Harry, and with Tyler. The stuff that was coming back from the production crew sounded great. The more people you get involved in making records, the more ideas you get, especially if the flow is good. Whereas when it all comes from one person it can start to sound flat. In this case, everyone was complementing each other, which gave the record depth and took it to a different place. I really believe in music as a communal thing, which is weird because I occasionally get some social anxiety, and my favourite thing is to make music on my own in my studio. But there's nothing better than making music with a group of people."

Styles' follow-up, *Fine Line*, was largely written and produced by Styles and Hull, with a lot of help from various people, mostly Tyler and drummer/guitarist Mitch Rowland, the four of them acting like a band. 'Watermelon Sugar' was one of several major hit singles taken from the album.

"'Watermelon Sugar' was the first song we wrote for the album. It was done in Tyler's studio in Nashville. I came up with a riff and Tyler put some chords to that, and Mitch jumped on the drums. We made a demo of the song, but our problem was to beat that demo. 'Watermelon Sugar' was really tough to get right. It taught Tyler and I how to produce, because we had to figure out what was great about the song, and there were problems with the demo that we could not fix, so we went through many different versions to get it right. It took 18 months to finish that song! So when we started work on *Harry's House*, we were like 'Let's get the demos as good as they can be,' so we could use them for the final productions."

> Kid Harpoon: "If a song sounds good on just acoustic guitar or piano, it's great. The problem is that sometimes when you write like that, it can get hard to produce it."

House Band

Harry's House was largely made by a small band consisting of Styles, Tyler and Hull, with help from engineer Jeremy Hatcher. The pandemic focused their minds in keeping the communal experience very contained.

"We started *Harry's House* when Covid hit. Harry couldn't go on tour, so he was like, 'Let's just go in the studio.' We went to Rick Rubin's Shangri-La Studios for a month, where we had first met Jeremy during the recordings for *Fine Line*. We turned the studio round, using the control room as a live room, and using the board in the live room. Tyler and I each had our own workstation, and Jeremy his recording setup.

"What was beautiful about it for me was that we were all worried, as in, 'Is this like the plague, are we all going to die? What the heck is going on?' The last thing on our minds was making a record. We were just friends together, hunkering down, having fun in the studio. We didn't really think about whether what we were doing was any good or not. That really translated into the music. I've honestly, hand on heart, never had as much fun as making that record in that studio. »

Kid Harpoon takes a surprisingly old-school approach to writing and production, preferring to work with real instruments and live performances.

» "With one song we did at Shangri-La, 'Late Night Talking', Harry came in with some slow, jazzy chords, and I suggested to do them faster. We then added some drums, and wrote some chords for the chorus section. We all just lean into each other's ideas. What's unique about the process with Harry, Tyler and I is that it really is just exploration. There is no judgement. Everything is up for grabs, no one's ever like 'That's a bad idea.' There's no aim, other than trying to achieve something that we love, and the only way of finding that is by exploring.

"The beauty of working with someone like Harry is that he's one of the best curators of people you'll meet. His taste is impeccable. Harry said, 'I trust each one of us and our tastes.' We write so many songs, and we keep listening to the good ones, while the ones that aren't as good sort of drift away. The beauty of my job in this situation is that I'm not making music for me, I'm making music to try and support the vision of the artist. So when we're working with Harry, Tyler and I are really feeling deeply what he wants, so we know that he'll like what we are doing."

Library Music

After a month at Shangri-La, the quartet disbanded. Other sessions took place

Harpoon House

Tom Hull's studio in Los Angeles, Harpoon House, naturally reflects his old-meets-new approach and his fondness for real instruments. "I basically turned the living room of my house into a studio. Which is a problem, because when the air conditioning switches on, all the electricity dips. I've got a ton of gear here. There's a whole drum setup, with Altec mic pres to record them for that vintage character. I use the Overstayer Modular Channel to get a bit of crunch on the drums. I also have a wall of drum machines, like [*Roland*] 909s, 808s, an [*Akai*] MPC3000, and a bunch of others. I'll play drums or I'll program a beat, and I'll run that through an Empirical Labs Fatso compressor or the Thermionic Culture The Rooster, or the UnFairchild 670M II, or something like that. On the way in, I like to get sounds as dialled in as possible, and then not touch them.

"I've got a couple of SSL G channels, recommended to me by Spike [*Stent*], because I wanted that SSL sound. I recently got the new SSL bus compressor, which is impressive. I actually own a 24-48 API board, but it's still in a box, as I'm potentially moving the studio. I love old reverbs and delays. I have my Lexicons, and Yamaha ones, like the Rev 7, which Spike told me is shit, but I think it's awesome. Plug-ins are amazing, and you can do anything with them, but there's something about sending stuff through outboard. Some electricity happens, and often you find something you weren't expecting. There's an element of discovery, and I find that this excites me the most. If you do something that you didn't know existed, you go like, 'Woah, I just did that!'

"I have so many keyboards and guitars it's a problem. Some favourites are an old 1960s Gibson Melody Maker, a really nice 12-string Guild, a Rickenbacker Tom Petty 12-string, and an incredible Gibson 335 that Harry bought for me. As for favourite synths, they include a Yamaha CS80, an ARP Omni and a Roland Jupiter-8. I have Pro Tools HD, and Revox B77, and Tascam eight-track tape recorders. I often record drums on them, before sending them to Pro Tools. Other bits and pieces include samplers like the Akai Professional S950, and the Ensoniq ASR-10, because those old samplers do the same thing as tape: they glue the sound. I went through a phase of buying a lot of old samplers, because they all do something different in the way they change the sound. Inphonik does a plug-in version of the 950 that's really useful.

"My Neumann U67 is my go-to when recording acoustic guitars. I also have great BAE Brent Averill 1073 mic pres, that I like to use on vocals. My vocal chain tends to be Telefunken ELA M 251, going into a [*Neve*] 1073, a [*Teletronix*] LA-2A and then into Pro Tools. But we recorded the vocals for Maggie Rogers' album [Surrender, *2022*] with Shure SM7s. We set up a posh microphone for her because we wanted to get the vocals sounding right, but later in the main room she said, 'Can I quickly chuck an idea down?' She grabbed an SM7, and just start ripping into it. I was like 'I hate the sound of this SM7, I feel like it's going to give us problems,' but the performance was just so much better, we had to figure it out how to make it work with the SM7, and we did."

later at Peter Gabriel's Real World Studios, joined by Mitch Rowland, and at Angelic and Henson Studios, in London and Hollywood respectively. But it was at the house of Rob Stringer, chairman of Sony Music Group and CEO of Sony Music Entertainment, that the biggest hit was conceived.

"Rob has a house out in the country west of London, in Henley-on-Thames. We hired a load of gear, and put gaffer tape everywhere. I was like, 'I really hope Rob doesn't kill us for this.' We turned the TV room into the main room, and did a bunch of songs there, including 'As It Was', 'Cinema' and 'Love Of My Life'. We did 'As It Was' in one day. Tyler came up with some chords, Harry started singing a melody, and I was like, 'Let's put a beat to that.' I'd also just gotten a Moog One synth, and all synth sounds on 'As It Was' came from that.

"We recorded live drums in Rob's library room, which was a real process. It has books everywhere, and we thought it sounded good, but in fact it sounded like trash. We had the beat, but with a really awful drum sound. So we separately recorded a kick, a snare and a hi-hat, and we placed each into the session, replacing the first drum part. We got a cool drum sound like that.

"We wanted the drums to feel more natural, so when we were at Real World we asked Mitch to play across the track. But we actually felt it lost something with the live drums, so we kept the drums recorded at Rob's place. However, at the end of the song the crazy fills Mitch was doing sounded awesome. So the drums in the first half of the track are a programmed, workaround kit, and then Mitch turns up and explodes the end of the song. It was a real build-it-together process. It's how we work."

Reference Material

Extensive work went into post-production and finishing *Harry's House*. "It is very much Tyler and I working in the box, and really getting into the details of it. We try to get the production and the rough mixes to the point where we're listening to them all the time, and Harry listens and gives his comments. Sometimes I'll do something else, and then do some work on my own, and Tyler will do the same, and we're always referencing each other.

"That's one thing Trevor [*Horn*] taught me when I worked with him: always have a reference of where you start. Whatever you do, start the day with a reference, and then as you work throughout the day, you can always go and listen to where you were at the beginning of the day, and listen to whether you have improved it. Maybe I've improved the drums and the bass but the vocals sound shit now, so I'm going to have to work on that. There's a lot of referencing old versions.

"I've tried to make the drums sound bigger, and technically managed, but found it's just not as good. They were better before. You don't know why, but they just had a feel. It's not about whether the drums sound good, because at the end of the day, people aren't listening to songs going 'Are the drums big enough, does the kick drum hit?' They're listening, going, 'Yeah, this is great,' because it's the combination of all of it. So we're constantly listening to the whole. It's like cooking: always taste the food. That's our finishing process.

"At the very end of the process we get into the Spike [*Stent*] world. Watching him mix is like watching David Beckham take a free kick. I like to get the rough mix to a place where I'm testing Spike, like 'Go on then, let's see how good you are,' and every time he knocks me out, it's insane. I go in there really confident, and he just makes me look bad at my job, because he just makes everything sound so much better!"

Bigger & Better

"Did I know that 'As It Was' would be as successful as it was? Genuinely, we had no idea that it was going to be as big as it was. You're aware you have a great song, but you have no idea what it's going to do. There's so much more to it. There are marketing campaigns, cultural timing and so much more. Everything has to line up in a certain way. Clearly everything aligned for 'As It Was' to hit big. But it's rare for all the stars to align like that.

"As part of the creative process, I find it very abstract thinking about a song commercially. When you're making a song, you have to think just about that song. You have to think purely about, 'Do I love it? Do you love it? What do we love about it? How does it make me feel?' Everything else is a by-product that you can't control. To even worry about that is to the detriment of something special and magical: making music."

RME Fireface UFX III

HUGH ROBJOHNS

USB 3 Audio Interface

Have RME made the perfect USB audio interface?

RME's Fireface UFX has long been the company's flagship interface, and I reviewed the very first model back in 2011. At the time, its 60 channels of simultaneous I/O (30 inputs and 30 outputs at base sample rates) was an outstanding feature, and it offered both USB 2 and Firewire 400 connections for hooking up to a computer. Such were the high technical standards of that original UFX that it was six years before its leading position was toppled — by the UFX+, introduced as part of the company's 20th anniversary celebrations.

In essence, the UFX+ was a UFX melded with a MADIface USB, expanding the original model's 60 channel I/O with 128 channels of MADI (AES10), delivering a whopping 188 simultaneous I/O channels — 94 in and 94 out! As the huge channel count was beyond the capability of USB 2, RME employed the novel USB 3 and Thunderbolt computer interfaces for the first time.

Six months later, in 2018, a 'next generation' UFX interface appeared — the UFX II — featuring upgraded converters but keeping the original I/O configuration. Indeed, the front and rear panel layouts remained virtually identical to the original UFX.

Introducing The UFX III

Now, five years later, I'm looking at RME's latest flagship multi-channel interface, the UFX III. This moniker appears logical enough, but the UFX III is not an updated UFX II. It's actually a UFX+ Mk2 — because it replicates the massive I/O architecture of the original UFX+ (which it replaces), while updating the analogue and conversion electronics with significantly better THD and SNR figures!

So, like all its UFX forebears, the UFX III has 12 analogue inputs (eight balanced line and four mic/ instrument), plus stereo AES3 and two sets of eight-channel ADAT — and, like the UFX+, it also has 64 channels of MADI, which my calculator totals to 94 inputs.

The output side is much the same, with 12 analogue outputs (eight balanced line and two stereo headphones), plus AES3, two sets of eight-channel ADAT, and 64 channels of MADI, making 94 channels again. Thus 188 I/O channels altogether.

For those that like making music for bats, and who don't need quite as many I/O channels, the UFX III supports sample rates up to 192kHz, but with a correspondingly reduced channel count. As you'd expect, each ADAT interface is limited to four channels at 96kHz and two at 192kHz, while the MADI I/O contracts to 32 and 16 channels, respectively. This means that at double sample rates the UFX III provides 54 channels in and out, while at quad sample rates it's reduced to 34 channels each way — all still mightily impressive. These counts assume all digital channels (ADAT, MADI, and AES) are synchronised to the same word clock; the UFX III cannot accept asynchronous digital sources.

Given such a vast channel count, computer connectivity is, once more, via USB 3, and to provide the most reliable connections RME has built in sophisticated error detection/analysis, displaying the results in the Settings menu. Any found errors warrant using a different cable and/or computer port, because RME have found these to be at the root of most USB 3 connection issues. In fact, the (superbly comprehensive, 128-page) manual provides very helpful compatibility information essentially advising the UFX III should only be connected to USB 3 sockets mounted directly on the computer motherboard (and using current chipsets from Intel, AMD, NEC/Renesas, and Fresco), and any remote USB 3 sockets connected to the motherboard via a cable are likely to introduce transmission errors, regardless of the chipset in use. So, you have been warned!

RME Fireface UFX III
£2727

PROS
- Superb converter and preamp performance.
- Immense connectivity in a familiar package.
- Triple AES and S/PDIF option.
- DURec now supports track naming.
- Integrated USB 3 error checking.
- Fully deserving of RME flagship status.

CONS
- None.

SUMMARY
The latest incarnation of the UFX platform boasts a stunning technical performance with vast I/O, familiar RME operation and features, and reliable USB 3 connectivity.

If all else fails, the UFX III also retains an option from the UFX+ which reduces the unit to a 30-channel USB 2 interface, dropping the MADI channels from the computer interface (although they are still accessible via TotalMix FX for monitoring and external routing).

For anyone who feels 188 channels still aren't enough, up to three UFX III units (or related Fireface units) can be used simultaneously (provided they are synchronised), appearing as a single combined unit. That said, TotalMix FX controls the DSP within each individual interface, and so channels from different hardware units cannot be shared for monitoring (unless via duplicated connections).

Core UFX Features

Anyone familiar with the earlier Fireface UFX models will feel instantly at home with the new UFX III, which looks and works pretty much identically. The new model confers all the familiar RME features we've come to love, including SteadyClock FS for solid, jitter-free clocking, DURec standalone multitrack recording onto a USB thumb drive (of up to 2TB capacity), dual five-pin MIDI I/O, Auto-Set gain controls, DC-coupled outputs (for passing CV or gate signals to modular synths etc), and class-compliant interfacing.

The latter has some caveats, though: a class-compliant USB 2 mode (without MADI) is available for all platforms (macOS 10.10 or later, iOS, Linux, and Windows 10-11), while class-compliant USB 3 currently works on macOS machines and some iOS configurations.

Although not strictly required for class-compliant operation, the bespoke USB 3 driver is available for both macOS and Windows and provides access to the interface's USB settings, such as buffer size, word-clock input termination, AES and MADI connection formats, AES/MADI/TOSlink output formats, clocking options, and so forth. This driver also installs the TotalMix FX app. For Windows platforms, there's a dedicated ASIO driver supporting buffer sizes down to 32 samples, which I was able to run reliably on my Windows PC.

For the UFX III the TotalMix FX app provides a powerful 4096-channel, 46-bit mixing engine with EQ, »

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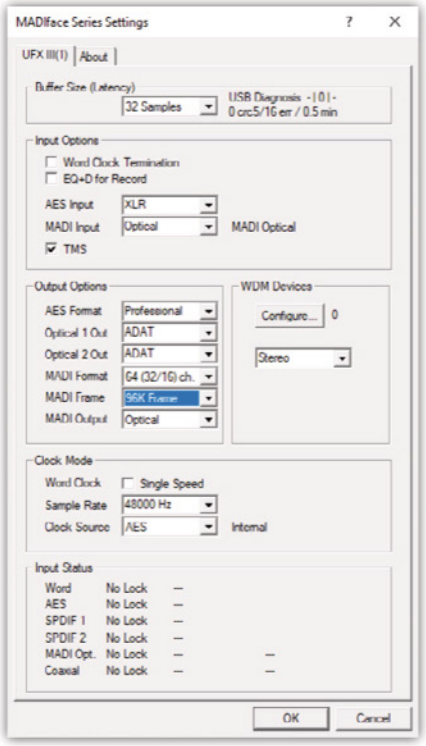
» dynamics and effects capabilities for comprehensive monitoring and external signal routing. As usual, TotalMix FX can also be controlled remotely via MIDI or through an OSC setup, as well as via the TotalMix Remote app. This can be installed on any PC, Mac or iPad connected to the same network as the host computer (via Ethernet or WiFi), greatly expanding the system's versatility and allowing user control away from the host computer.

RME's optional ARC USB remote panel can also be used to control the UFX III, either via a USB socket at the unit's rear, or via any USB port on the host computer, and the ever-useful DIGICheck Analyser app is also available, of course. This provides comprehensive level metering in multiple formats including sample peak, RMS, true peak, BS1770 Loudness, and K-system. A correlation (phase) meter, stereo and surround vectorscopes, bit statistics, channel status and spectral analysis are also included, all driven from the UFX's DSP.



The UFX III's control panel provides options for the buffer size, digital input and output formats, clocking source, and sample rate. Alongside the buffer box is the USB diagnosis report, which shows errors. This image was obtained by plugging into a front (cabled) USB 3 port. Connecting to the rear motherboard port gave zero errors reliably.

Hardware

As with previous incarnations, the UFX III is housed in an elegant 1U rackmount chassis but, while it outwardly appears the same as its predecessors, the innards have been completely redesigned, with new converters and mic preamps that are essentially the same as those used in the outstanding 12Mic.

Four combi XLRs on the front panel accept mic or instrument inputs, and a pair of quarter-inch TRS sockets provide two independent headphone outputs (but can also be used as unbalanced line or CV/gate outputs). Next along are a pair of five-pin MIDI in/out sockets, and then a USB 2 Type-A port for DURec recording to a thumb drive (which supports track naming, as in the UCX II). The remaining right-hand third of the panel is the control section, with a small colour screen which shows I/O meters when not used for configuration. The menu operation is identical to the UFX II interface, and similar to that employed on the UCX II and the ADI series, so will be plain sailing to most RME users. Status LEDs provide clocking information on the left column and MIDI monitoring on the right and, as in the UCX II, the front-panel controls can be remapped to suit specific standalone operations as well as locked to prevent accidental operation, if required.

The rear panel is equally familiar, starting with octets of balanced line inputs and outputs on TRS sockets — except outputs 1+2 which are on XLRs. AES3 in/out are also connected via XLRs, while two sets of ADAT TOSlink ports are provided, with both sets being reconfigurable as optical S/PDIF if required, making three AES or S/PDIF channels available.

A pair of BNC connectors provides word-clock in/out by default but can be repurposed as a coaxial MADI I/O, if necessary, although the primary MADI interface is on standard SC fibre-optic ports. Having both fibre and coaxial MADI options allows for redundant connections, and the UFX III switches automatically (in less than a sample) between the MADI optical and BNC inputs if the active source is lost. MADI outputs can also be mirrored to both the optical and BNC connections, for redundancy again, if required.

Supplementing the front-panel MIDI I/O is a second set of five-pin MIDI in/out sockets at the rear and, like other RME units, MIDI data can also be sent and received via the MADI connections. The UFX III has an internal (Class 1, grounded) universal power supply accepting any voltage between 100-240 V AC through a standard IEC inlet (consuming between 19 and 25 Watts). The host computer is connected via a 'double-decker' USB 3 Type-B socket, and both USB 3/USB-C and USB 3/USB-A cables are provided in the box (along with a mains cable and ADAT light-pipe fibre). Finally, a USB 2 Type-A socket accepts the optional ARC USB remote controller.

Delving into the analogue I/O technicalities for a moment, alignment options are similar to other RME units, with inputs being switchable to achieve 0dBFS at either +13dBu or +19dBu (equivalent to the +4 and Lo Gain modes in legacy RME interfaces). On the output side, 0dBFS generates either +19dBu, +13dBu, or +2dBV (+4dBu) at the TRS outputs, while the XLR outputs (channels 1+2) have a fourth option delivering +24dBu.

It won't bother most potential UFX III users, but it's a shame that the UFX III doesn't support +24dBu peak level on all its line I/O — especially when other recent RME interfaces do — as it potentially complicates connecting the UFX III with professional mixing consoles and outboard equipment working with 20dB of headroom.

Headphone outputs on the front panel use the same D-A converters as the balanced line outputs and can be switched between High (+19dBu) and Lo (+4dBu) operating levels in a similar

Round the back things are pretty busy, but not as busy as you might think with so much potential I/O.

way. However, their output stages are optimised to drive low-impedance loads, delivering maximum power (200mW) into 32Ω headphones. All 12 analogue outputs are DC-coupled, avoiding LF phase shifts for audio signals, and allowing the generation of DC control signals (CV and gates) for synthesizer modules or for scientific measuring equipment. When set to the High output mode, a maximum DC voltage of +9.5V can be generated, reducing to +4.75 for the +13dBu setting, and +1.7V for the Lo (+2dBV) mode.

The four microphone preamps (channels 9-12) offer a gain range of +8 to +75 dB in 1dB increments, plus a unity gain (0dB) setting allowing the XLR to accept line inputs (up to +18dBu). The preamp's EIN figure is a very impressive -129.8dBu (60dB gain, 150Ω source, A-weighted) — meaning these are very quiet preamps! If selected, the AutoSet feature optimises the preamp gain automatically to preserve 6dB of headroom.

Phantom power is available on the XLRs, of course, but doesn't reach the combi-jack instrument connections, which are presented with the usual 1MΩ input impedance. For instrument inputs the preamp's maximum gain is reduced to +50dB and can deliver 0dBFS for any signal level between -21 and +21dBu. As you'd expect, the technical specifications for all inputs is exemplary, with the very worst THD figure being 0.00045 percent and the best 0.0001 percent! My own Audio Precision bench-test gave AES17 dynamic range figures of 119.1dB (A-weighted) for both the A-D and D-A converters — which is hugely impressive.

In my personal league table of measured converters, the UFX III's D-A performance places it between the Focusrite RedNet or Crookwood M1 mastering console and the Grace Design m905 or Lynx HiLo. And the A-D performance lies between Prism Sound's Dream ADA128 and Cranborne's R500 — so this is a genuinely top-notch, mastering-grade interface!

Bottom Line

For readers who habitually skip to the end of a review, the bottom line is that the new UFX III is much the same as the old UFX+, but with seriously impressive converters and a quartet of stunningly clean and noise-free mic preamps — plus all the wealth of connectivity and features we've come to associate with the UFX marque. The new TotalMix Remote app will be a boon to many, too. I couldn't fault the UFX III in any way and it's impossible not to be impressed with RME's latest offering, which clearly deserves its flagship title.

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Sonible smart:gate

AI-assisted Noise Gate Plug-in

Just why would you make the humble noise gate 'smart'? Sonible asked that question, and this plug-in answers it!

JOHN WALDEN

Sonible were pioneers in the use of AI and machine learning in DAW plug-ins, and in recent years they've delivered an impressive range of processors that make use of that technology. Their latest offering is smart:gate and, as might now be expected from the 'smart' series, this looks to apply the benefits of AI technology to the noise gate.

A traditional noise gate essentially opens (to let audio through) and closes (to block it) in response to the incoming signal level — the user can usually set the threshold at which this open/close state changes. Most modern gates, whether hardware or software, also allow the user to control details such as how quickly the gate opens, the minimum time during which the gate remains open and the opacity or floor of the gate when closed (if it's only part-closed, it brings the signal level down to that floor). Of course, smart:gate can be used in this fashion — in fact, as a conventional gate, it's very well specified — but of course this gate is also smart. So let's see what else it can do...

Sonible smart:gate
€89

PROS
- Clever — and very useful — content aware gating.
- Easy to use...
- But plenty of detailed control available.

CONS
- It's not magic; like a conventional gate, this is not an unmixing process.

SUMMARY

Sonible's smart:gate fully justifies its name. A very clever application of AI that brings some additional 'smarts' to the routine task of gating noise.

Context Is Everything

One thing smart:gate offers that's out of the ordinary is a 'content aware' mode. As shown in the main screenshot, Sonible provide a number of common target profiles, and once you've selected one of these the plug-in analyses a short section of the incoming audio. It attempts to identify, within that input signal, a match for the selected target and then configure initial settings — once done, the gate should open only when the level of the selected target exceeds the threshold. Other sound elements within the signal will not trigger the gate. And for many sources, this might be all that's required.

As with all Sonible's smart plug-ins, the UI provides plenty of graphical feedback that enables you to visualise exactly what the gate is doing. When working in its content aware mode, this includes identifying exactly which parts of the input signal's waveform are identified as a match for the selected target, as well as the state of the gate as it changes over time. If you feel the need to fine-tune the conventional gate controls (attack, threshold, release and hold, for example), this visual feedback is particularly helpful; those controls operate exactly as you'd expect, but only on the selected sound. It's a very neat trick and in general it works very well.

Bells & Whistles

Of course, while Sonible's AI assistance can do the heavy lifting, you might still wish to »

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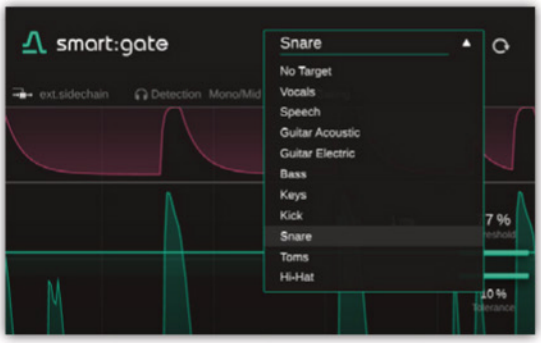


smart:gate analyses the incoming signal for the information that most closely resembles the chosen target profile — it will only open when it detects those sounds in a signal.

» dig a little deeper, in which case there are plenty more options you can explore. For example, with the gate already targeting only a specific sound type, the Level Bias control lets you change the emphasis placed on input level of that source in controlling the gate. A low Level Bias setting can suit sounds without a particularly strong transient (eg. vocals), while a high Level Bias lets the gate follow sounds with stronger transients (such as drums) more effectively. The plug-in offers a Mono/Mid vs Side switch, to change how the detection process responds when using a stereo input, and you can also switch between gating and ducking modes. The latter can be used with the side-chain input and, for example, would make it very easy to duck the level of a backing track using a send from a voiceover into the side-chain.

Perhaps the most interesting addition is the Impact Controls. These can be adjusted at the base of the UI and a full panel can be popped open to provide a more graphical tweaking experience. The Impact slider controls the degree to which the gate actually shuts when the target signal isn't present, with 100 percent being fully shut and 0 being fully open (essentially, the same as bypassing the plug-in). While this capability is found on lots of noise gate designs, the three further controls (band suppression for the low, mid and high frequencies) are not. These essentially operate to control the release time of the gate in each frequency band (higher values force faster release times) and you can also adjust the crossover points between the three bands. These controls require a little familiarisation time, but, usefully, you can solo a band to hear what's going on and they do provide some very interesting ways to finesse the tonality of your signal.

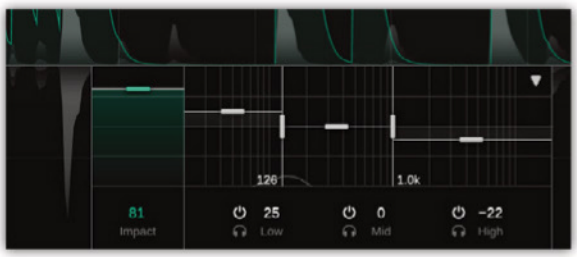
Finally, amongst the other features, it's worth noting the eight State slots at the top of the UI. These let you save different settings in a single instance of the plug-in and easily toggle between them as you fine-tune your settings. All the plug-in's key parameters are available for automation, but not these — it would be very cool if you could also automate the State number, to quickly switch between full configurations that suit different parts of a performance.

Smart By Name, Smart By Nature?

For those who deal with a lot of multi-channel acoustic drum recordings, smart:gate could probably pay for itself very quickly in time saved. It can be tricky when miking a kit not to end up with significant bleed on your main close mics. That can mean, for example, that your kick is pretty loud on your snare recording and *vice versa*, and that can leave you with less control than you'd like. With its content aware approach, smart:gate offers the potential to isolate each drum very effectively.

I realise, of course, that a little bleed can often help to give a drum recording its sense of coherence — you won't always want to remove it entirely — but with better isolation of your kick, snare and hat sounds, it's so much easier to dial in your EQ and compression on those elements. Whether you want to rework the sound in that way before blending in your overheads to add back a sense of 'a kit in a room', or to start with your overhead image and then shape the close mics (or use them as triggers) to reinforce that overhead image, smart:gate can be a very effective tool. Also, it's a fairly painless process: though there's scope for tweaking, I found that the initial AI-suggested configuration was generally pretty close to what I wanted.

It's worth me stressing that smart:gate is still operating as a gate, by which I mean it's not performing any sort of 'unmixing' process to do it's thing. This means your audio is left intact, without any nasty artefacts, but in gating your kick, if your snare plays at different times to the kick then it will easily be kept out by smart:gate's content aware processing — it will identify only the kick hits and not the snares — but if a snare hit coincides with a kick hit, as the gate opens for the kick it will also allow the snare to bleed through.



The Level Bias facility offers a means of fine-tuning the gate's response to suit different sound sources.

I've talked a lot about drums because it's such a common application for gates, but they can be used on other sources too, and one application I explored during the review period was cleaning up vocal recordings — trying to remove backing track bleed from headphones. With the same qualifier that I noted just above, smart:gate did very will indeed at this job.

A final thing to point out is that smart:gate's smart gating comes at the cost of a little processing latency. So while it works very well as a mixing or pre-mix prep process, it's not really appropriate for use in live sound applications.

Get Smart?

I find Sonible's smart plug-in range very impressive generally, and smart:gate is no exception — the 'smart' label is definitely justified, as it enables you to approach the task of gating in a way that you just couldn't with a conventional gate. Importantly, it can be very easy to use: pick the appropriate target source, let the plug-in analyse the incoming audio and it's a case of 'hey presto! Job done!' But I love that you can then dig into the details should you feel the need.

So, if your music production work regularly involves using noise gates — the most obvious scenarios are with multi-channel drum recordings or full band tracking sessions – then smart:gate could make it much quicker and easier for you to clean up unwanted bleed captured on different mics, and in a commercial setting where time means money there's a lot to be said for that. Helpfully, there's a fully functional 30-day free trial, and it's well worth downloading.

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Extinct Audio BoRbon

Ribbon Microphone

It might not look like a classic ribbon mic, but there's a lot more to Extinct Audio's BoRbon than meets the eye.

SAM INGLIS

The words 'ribbon microphone' conjure up the titans of the mono age: the Coles 4038, BBC Marconi Type A, Melodium M42B and, perhaps most iconically, the RCA 44. But there were also many smaller, more humble designs that found their way into the hands of home recording enthusiasts and small-time PA operators. In the UK alone, manufacturers such as Reslo, Grampian, Film Industries, Cadenza and Selmer produced countless ribbon mics of varying cost and quality.

There's probably no-one alive who knows more about these vintage British ribbons than Stewart Tavener of Xaudia. As well as repairing thousands of them, he's developed a standard set of upgrades for the popular Reslo RB mics, based on an old BBC research paper. Having also accumulated a pile of spare bodies and grilles from unrepairable mics, he began to wonder whether there might be mileage in retrofitting new ribbon motors into these shells to get them going again.

Stewart already made ribbon motors as part of his other business, Extinct Audio, but Extinct's existing BM9 microphone is a so-called 'long ribbon' design, and its motor would be too large to fit into Reslos and other antique British models. Consequently, a new 'short ribbon' motor was developed, and this turned out to perform so well in testing that the idea of using at as the basis for a new Extinct model took root.

The BoRbon Dynasty

Although the BM9 is very competitively priced for an entirely British-made microphone, it's still a big investment for many home studio owners. That's par for the course for mics made in Western or high-wage economies, and models such as the beyerdynamic M 130, Coles 4030L, Audio-Technica AT4080 and Rode NTR are all comparable in price. If your budget won't stretch that far, there are impressive 'boutique' offerings from Hohm and Alder Audio that come in a couple of hundred cheaper, but they have to be ordered from Australia and the USA respectively. Below that, you're looking at Chinese-made mics, which are mostly based around one or two generic designs, and don't always offer the same levels of quality or consistency. With their new mic, Stewart and his team at Extinct Audio wanted to see if they could find a way to rival these imports for affordability.

Extinct Audio BoRbon

£425

PROS

- The sound of a classic long-ribbon microphone in a small, lightweight package.
- Unique looks.
- Remarkably affordable for a UK-made mic.

CONS

- Figure-8 pattern isn't perfectly symmetrical.

SUMMARY

Looks a bit like a biscuit. Sounds remarkably like an RCA 44!

One of the major factors in the cost of microphones is the bodywork. Casting and finishing precisely shaped metal shells is an unavoidably expensive business. Some mic manufacturers have reduced costs by outsourcing this work overseas and keeping the more important stuff in-house, but Extinct didn't want to go down that route. Instead, they came up with an impressive piece of lateral thinking. They were already using a local laser-cutting company to etch the Extinct logos into wooden boxes for the BM9. What if the microphone itself could be made from wood rather than metal?

The first prototype convinced them that the idea would work, and the result is the BoRbon, so-called either because it's a 'beau ribbon', or because it looks a bit like a Bourbon biscuit. I don't know that you'd bite into one by mistake, but in the flesh, it certainly resembles a food item more closely than it does any other mic. The look and feel are utterly unique: in essence, the body is built up from five layers of laser-cut ply, sandwiched between shiny plastic faceplates. Three pairs of machine screws run all the way through the body. The top two pairs pass through the corners of the ribbon motor, holding it in place inside the wooden frame, while the larger pair at the bottom attach an external right-angle XLR connector. The review mics were finished in a very classy tortoiseshell, but a bonus of this construction method is that Extinct can offer custom colours and designs. At the time of writing, for example, they're offering a fetching gold flake version as a limited edition.

As well as the potential for customisation, the BoRbon design has several other benefits. First, it's incredibly compact. Of the other ribbon mics in production today, only the beyerdynamic M 130 is comparable in size; the BoRbon is wider and deeper than the M 130, but is even shorter. Second, you can place two of them in close proximity without having them jump towards one another. Both of these factors are plus points for stereo use, allowing you to easily arrange two BoRbons in a truly coincident arrangement. Third, it weighs very little. This is good, because its unusual design means that conventional mic clips won't accommodate it. Manufacturing a custom shockmount would have pushed the price up, so instead, Extinct have developed an ingenious new version of their Mount-A-Mic (see box). You probably wouldn't want to use this with an RCA 44, but it's perfect for a featherweight like the BoRbon. This is supplied along with a leatherette pouch that holds the mic snugly; again, a wooden box would have added to the cost, so this seems a perfectly reasonable compromise.

Mount-A-Mic Mk2

Under his Xaudia brand, Stewart Tavener created the Mount-A-Mic back in 2015. The original version is a metal barrel that has a female XLR connector at one end and a 5/8-inch threaded socket at the other, with the mic cable emerging from a hole in the side. It's a simple, effective and compact way of attaching a mic to a stand in circumstances where you don't need shockmounting and you're happy to mount the mic pointing upwards.

The new Mount-A-Mic Mk2 is equally ingenious but much cheaper to manufacture. In essence, it consists of a single small piece of ply with two circular holes laser-cut into it. The smaller of the two holes is 3/8ths of an inch in diameter, so you can slide it directly onto the business end of a mic stand and secure it with a threaded washer, or attach it to the included 5/8 to 3/8 thread adaptor. The larger hole is shaped precisely to fit between the metal barrel and the plastic cable gland on a female Neutrik XLR connector. Simply unscrew the plastic part, pop off the barrel, put the Mount-A-Mic in between and reassemble the connector, which takes less than a minute. It's perfect for a small, lightweight mic like the BoRbon, and is supplied along with a short cable. A stereo version for mounting two BoRbons in a Blumlein array is also available.

Shapes & Sizes

In the world of capacitor mics, large-diaphragm and small-diaphragm capsules are distinct designs. You couldn't create a KM84 capsule by scaling down a U87, or *vice versa*. By contrast, there's no fundamental difference between long and short ribbon motors. All else being equal, making the ribbon longer increases sensitivity and signal-to-noise ratio; but sensitivity is also related to the strength of the magnetic field within which the ribbon moves. Longer ribbons need to be wider than short ones in order to retain mechanical strength and stiffness, and this in turn pushes the pole pieces either side of the ribbon further apart, reducing the strength of the field. The upshot is that the length of the ribbon isn't the main factor determining the technical performance of a ribbon motor. With today's powerful rare-earth magnets, long and short ribbons can perform more or less equally well as long as they are equally well designed. The BoRbon bears this out to a surprising extent, as we'll see.

"The BoRbon could be a lot heavier than it is, and it would still punch a long way above its weight."

Apart from an output impedance of 250Ω at 1kHz, formal specifications weren't available at the time of writing, but I would expect its measured performance to be typical of a well-designed passive ribbon mic. In a direct comparison, it seems a couple of dB less sensitive than my BM9s, but still puts out a healthy level. Like almost all ribbon mics, the BoRbon has a figure-8 pickup pattern, but unlike most, its internal construction isn't quite symmetrical front-to-back. In some designs, such as the Royer R121, this asymmetry is intentional, and is meant to give the front and rear of the mic different tonal qualities. In the case of the BoRbon it's just a by-product of the design; it might be an issue if you want to use it as the Sides mic in an M-S array, but the »

» front and back sounded pretty similar to me.

Unlike the BM9, the BoRbon is not offered as a matched pair, but if you ask Extinct nicely, they'll try to pick out two similar examples for you. The two BoRbons supplied for review were indistinguishable tonally, but one was perhaps 1dB more sensitive than the other.

Vintage BoRbon

It's always useful to compare new mics to a known quantity when you're trying to get a handle on how they sound, so my first tests involved capturing the same sources on the BoRbons and on my own BM9s. This showed that the two Extinct mics have clearly different characters. Rather like the Royer mics, the BM9 has a significant presence boost in the upper midrange; it could even be called 'bright', or at least bright for a ribbon mic. And, as you'd expect of a figure-8 mic, it also exhibits a strong proximity boost when used up close. The BoRbon demonstrates an equally prominent bass tip-up, but the BM9's midrange presence boost is completely absent. Consequently, although it clearly has useful response well beyond 10kHz, it sounds quite a bit darker and smoother than the BM9s and Royers of this world.

The surprise to which I referred earlier came about in my next comparison. By chance, the BoRbons arrived for review at the same time as the Cloud 44 Midnight: a classy 'long ribbon' microphone which is marketed as an updated version of the iconic RCA 44. Out of curiosity, I put the two mics side by side on a stereo bar and sang into them — and they sounded uncannily alike, to the point where I'm fairly sure I couldn't have confidently picked the raw files from both mics apart in a blind test. It was only after I'd applied some treble boost and a huge amount of compression that differences began to make themselves apparent, and even then, these were slight. The Cloud had slightly more going on in the 5kHz area, and its overall sound was perhaps a touch smoother, allowing you to go further with EQ before any roughness becomes apparent. But you could absolutely use either of these mics to drop in on a vocal recording made with the other, and I'd be amazed if anyone noticed.

Of course, on-axis pickup is only one aspect of a mic's performance, and in fairness, the difference was a little more apparent on drums, where the Cloud had greater bass extension and a marginally more polished-sounding midrange. Even so, it was still a close-run thing. The BoRbon could be a lot heavier than it is, and it would still punch a long way above its weight.

Because of its asymmetrical design, the rear side of the BoRbon may sound a little different from the front.

Outside In

Engineers and musicians alike are often guilty of judging things with our eyes rather than our ears. We 'know' that large-diaphragm capacitor mics are good for studio vocals, so we feel uneasy using a pencil mic, though there's no technical reason why we shouldn't. And we associate the classic 'long ribbon' tone with the mics that made it famous, most notably the RCA 44. The BoRbon is tiny and weighs next to nothing, which is a pretty important consideration if you're using it as a drum overhead or similar. It looks nothing like the RCA 44, or indeed any other mic. Yet it absolutely nails that sound. And it's half the price of any comparable UK- or European-made ribbon mic. Musicians might need some convincing to take seriously a mic that looks like a biscuit — but once they've heard the BoRbon, they'll be in no doubt.

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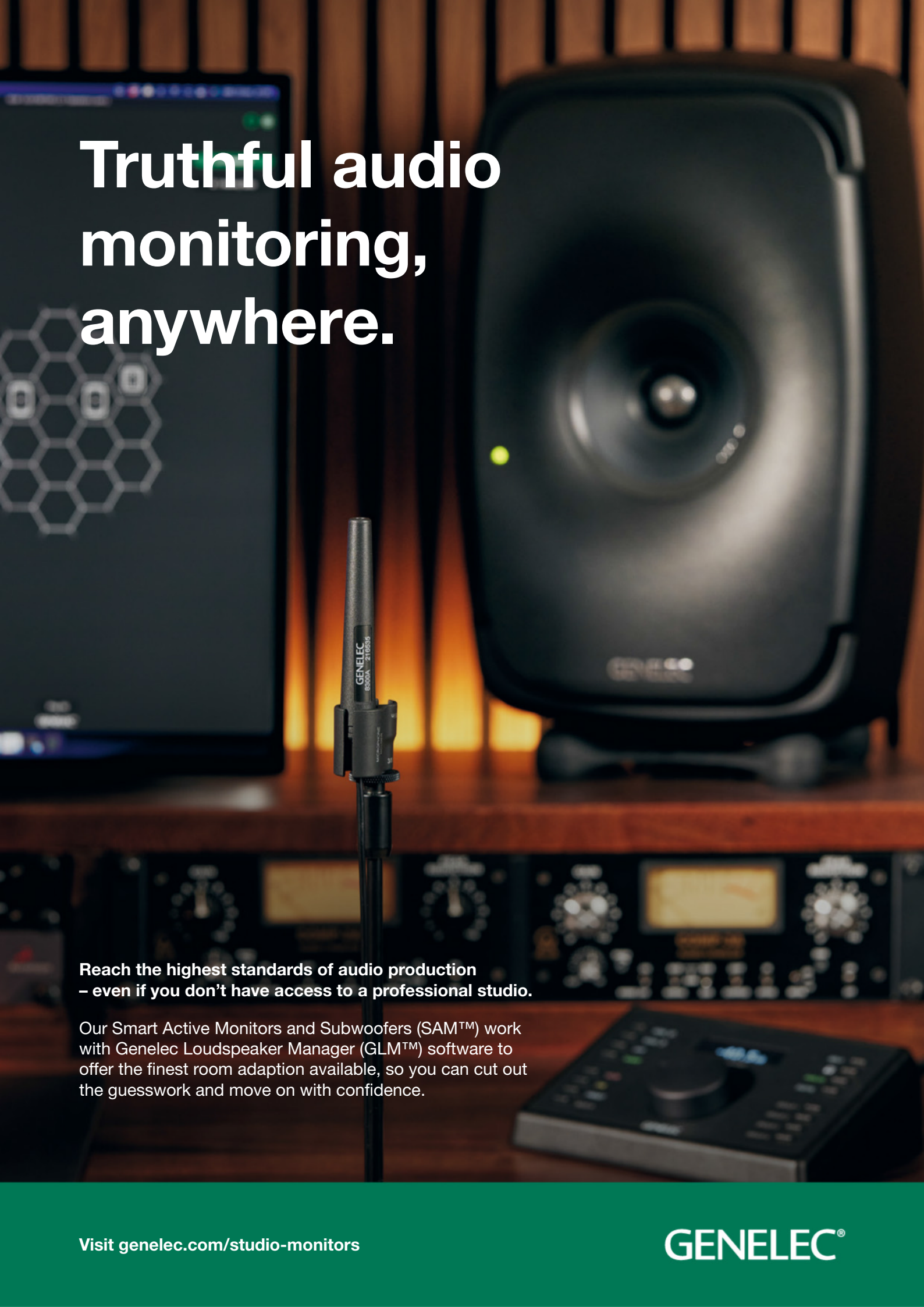

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ROBIN FREEMAN

When *SOS* asked if I might like to share my 'cheat sheet' more widely, I was happy to oblige. It came about thanks to my role teaching the History Of Production course at the Abbey Road Institute in Amsterdam where, as well as delivering the standard course content, I've always been encouraged to supply tips and anecdotes from my personal studio experience. This approach went down well with students but a few years ago one of them asked if I had "anything on paper." As you can imagine, it's pretty hard to distil the knowledge gained over nearly 50 years in the studio into a few short tips. But I gave it a go and handed out a sort of 'stream-of-consciousness' list of many tips to my class. It met with some very positive reactions, and I hope you find something useful here too.

1. Performance First

The artist's performance is far more important than any technical aspect of the recording. Whether engineering or producing, it is your responsibility to create conditions in which the artist will be able give their best performance.

2. Plan & Prep

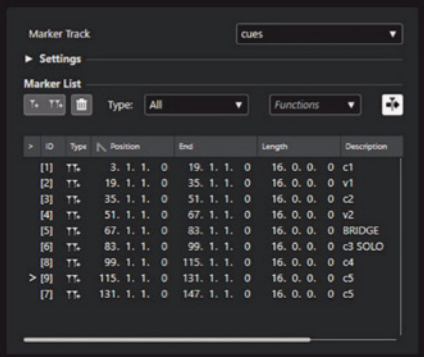
With tip 1 in mind, prepare things in advance of the session where possible. The client does not want to wait while you get your technical stuff together.

3. Do Not Bluff

If something technical is wrong then let the client know, as if you don't it will probably come back to bite you. (Not everyone agrees with me on this one!)

4. Know Your Place

Know where you are in the song. Make use of markers or cue points, so that if someone says "let's hear it from the second chorus" you will be there right away.



The importance of quick navigation using markers is too often overlooked.

5. Set Phase To Stun

All-pass filters can adjust the phase of a signal without changing the frequency response.

The more mics you use to record a sound source, the more likely it is that phase cancellation problems will arise. A case in point is recording a drum kit, where multiple mics will capture the same sound sources from different angles and distances (the overheads versus the close mics, for example). By adjusting the phase relationship of the various mics, you will be able to achieve a fuller and more satisfying sound. Do this by adjusting mic positions where possible. After recording, you can compensate to some extent by micro-shifting the various drum tracks in time (zoom in on the waveforms when doing this), or using a phase alignment tool (an all-pass filter) such as Little Labs' IBP, which is available as a Universal Audio plug-in.

6. It's A Pass

A 'bracket' EQ, with both high- and low-pass filters band-limiting a signal, can be a useful tactic — with good judgement.

Do use 'bracketing EQ', meaning high- and low-pass filters, to reduce the volume of the parts of the frequency spectrum *that are not needed for the instrument in question.* The bit I italicised is crucial — I didn't say "high-pass everything except the bass and kick," which is a common mistake. Try fairly soft filter slopes at first, perhaps 12dB/octave, as they generally sound more musical than steep slopes. Or try using gentle shelving EQ instead.

7. A Place For Everything

When mixing (and writing/arranging) try to ensure that each instrument has its own space in the audio frequency spectrum. Panning similar sounds away from each other may seem like it fixes a masking problem, but if you get it right in mono using EQ, the stereo mix will be fine too and you will be in better shape. It took me many years to learn this lesson.

8. Beware Treble Boosts

Don't use equalisation to boost treble on *everything*. If you want the song to sound brighter, try using a bright instrument.

9. EQ In Context

Try not to adjust EQs when in solo mode, because if a sound is unpleasant in isolation but great in the mix that's fine. The sound in context (the whole mix) is what counts.

10. Near & Far

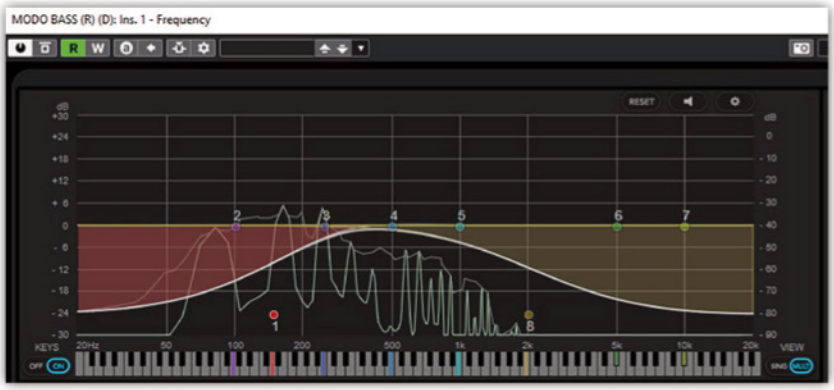
Sounds with more reverb and less high-frequency content will appear to be further away. Dryer and brighter brings a sound to the foreground.

11. Where's The Bass?

If your bass sounds fine on its own but gets lost in the mix, it's probably some other instrument that's to blame — something else is masking the bass. First do your best to find the culprit and EQ out any offending frequencies. When you've done that, you can often also give the bass extra definition by adding a little distortion, particularly to the mid and higher frequencies (this will generate extra harmonics), and perhaps try an EQ boost around 1kHz.

12. Compression

Do not over-compress unless you are sure it's the right thing to do. These days, dynamics are the name of the game. »



If you can't hear the bass on small speakers, you don't need more lows, you need more presence in the mids. Distortion can help with that.

When setting up compression, try a slow attack and fast release as a starting point.

13. Compressor Timing

Regarding compression, there are good reasons to use different attack and release times, but I generally start with a slow attack (to preserve transients) and a fast release, and see how that sounds. When using parallel compression, you can be much more aggressive with a fast attack time and lots of gain reduction: start with the dry signal and mix in to taste.

14. Clip Gain

Consider adjusting your recorded levels with edits or automation before using any dynamics processing. For example, on a vocal track, adjust the levels per word or phrase to be close to the way you want them — raise the level of soft passages and drop the level of very loud bits — and create broadly the right balance before the part hits a compressor. Also consider reducing the level of excessive breathing between words and even de-essing by changing the clip gain on sibilants before further processing.

15. Dynamic Balance

Add interest to your mixes by making the balance dynamic, for example introducing changes with each song section. And don't forget to ride your effect returns: it's very common to have vocal and instrumental effects such as reverb and echo constantly move up and down to suit the moment.

16. Rest In Peace

Take regular breaks, to give your ears a rest and keep things in perspective.

17. Referencing

Listen to reference music to make sure that you are in the right ball-park regarding the overall sound: bass, treble, compression etc. But don't be fooled by the fact that your references have been mastered and will probably be brighter and louder than what you are working on.

18. Listen Like An Amateur

After developing the ability to listen analytically, you'll probably need to re-learn how the general public hears music: as a block of sound that causes them to feel a particular emotion. Make an effort to do that, as the average listener is not interested in your snare sound or how much 2kHz there is in the guitar, only in the way the music makes them feel.

19. The Mix Bus

You don't have to add stuff to your mix bus. These days I generally use a little EQ and perhaps a little compression. But I made records for years with nothing on the mix bus. More can always be added when mastering. (And please, do not *limit* the mix bus because that ties the hands of the mastering engineer.)

20. Two Little

Often, two devices arranged in series, each working just a little bit, will sound better than one device working harder. This can apply to both compressors and EQs, for example.

21. Work The Cue Mix

The musician's headphone mix is of critical importance and can make the difference between a great and a poor performance. Spend time making sure the artist is hearing exactly what he/she wants to hear. You can manipulate a singer's distance from the mic to a certain extent by giving them more or less of themselves in the cans. You generally don't want your singer to get too close to the mic; be aware of the proximity effect, which is a rise in bass response when close to a directional mic (eg. cardioid).

What you put in an artist's headphones matters.

22. Beware Bass Boosts

In general, be wary of boosting bass frequencies, as these eat up a lot of energy in the mix and can't be heard on some playback systems. If in doubt, check against known reference tracks.

23. Learn The Rules To Break

Being different is good, and breaking rules can lead to originality. But try to be aware why the rules exist, and why you are breaking them.

24. Take It Too Far, Then Back Off

When setting up plug-ins or outboard equipment, try extreme settings to clearly hear what the device is doing, and then back off to a more suitable amount of processing. Be aware that some processing, such as harmonic enhancers ('exciters'), can be addictive, and after applying more than is required your ears will need a chance to 'get back to normal' before you're able to make a good judgement again. Specifically with exciters, a little less than you think is enough will generally be just right.

Beware! Your ears can grow accustomed to the sound of harmonic enhancers, making it easy to overcook things.

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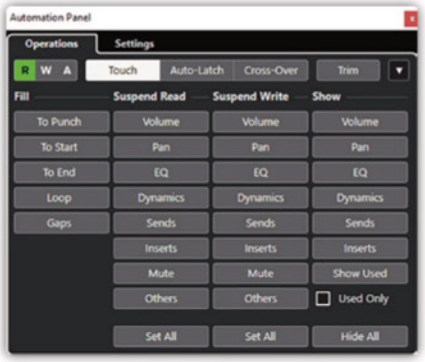
» 25. Automation Rules



Automation is a powerful tool for much more than levels — it's worth learning your DAW's system in depth.

Become very familiar with your DAW's automation features. In the days before automated mixing existed, the engineer, the producer, the whole band and maybe a few roadies too all had their hands on the console at mixdown. I now like to end up with nothing manual to do during a final mix, so everything is automated. But avoid the temptation to automate the soul out of the music. An occasional unexpected level jump or extreme effect setting can often be a good thing. Do your 'human' balance passes and have the computer repeat them for you. You can then tweak to taste — but don't tweak too much.

26. The Master Fader

You may find it useful to 'ride' the master fader so that your choruses are maybe 1dB louder than your verses, for example. There is no law that says the master fader must remain static throughout the song.

27. Commit To Decisions

Don't postpone too many decisions while tracking: 200-plus tracks at mixdown time will not lead to a good end result. So try to make choices along the way and commit to them. You can always save the ingredients of a pre-mix so that you can redo it later if absolutely necessary. I generally have a muted folder in my project called Parking, in which I save all these ingredients. (I also disable its contents to save on processor power.)

28. Comping

Back in the day we would often record a performance in little pieces, punching in to fix things as needed. In later years, I realised that it is better to capture longer performances, in complete takes or large sections, and then 'comp' the best bits together. This approach generally gives the most natural feel and flow to the performance and is less stressful for the artist too.

29. Embracing Spill

As far as possible, have your musicians play together. This will generally give you a better result than recording each player separately. Assuming your players are competent and don't make mistakes the whole time, don't worry too much about spill between microphones (eg. drums in the piano mics or guitars in the drum mics). In a good acoustic space, the leakage may even enhance the sound of your recordings. The late, great Al Schmitt had something to say about this, the use of the omnidirectional mic pickup pattern and much more in his books *Al Schmitt On The Record* and *Al Schmitt On Vocal And Instrumental Recording Techniques*. Recommended reading.

30. Find Your Own Way

Experiment and try to find new ways of doing things. Being able to copy someone else's technique can be very useful and educational but the most original and rewarding results will come from striking out on your own and pioneering a new approach. Easier said than done, but worth striving for.

31. Listening Levels

A good reference monitoring level will help you make better, more consistent judgements.

Don't fool yourself by listening too loud or too soft. When you listen very loud, your ears work like compressors. Most stuff sounds better when you play it loud but it's really hard to judge balance that way. Find a monitoring level between those two extremes and stick to it most of the time. If the band wants to listen to a playback of their performance at rock concert volume, by all means oblige — but leave the room while they're listening to protect your hearing.

32. Keep It Real

For 'real' instruments, listen to the way the instrument sounds in the studio first, then (assuming it sounds good!) see if you can capture that sound in the control room. Moving a microphone a few centimetres can make a big difference — it is really useful if you can have an assistant do that for you in the live room while you listen to your control room monitors.

33. Quiet, Please!

A quiet listening check on small speakers can shine a helpful light on the midrange.

Listening very quietly indeed, in mono on a small speaker (like an Auratone) is a step I generally take during the mixdown process. You will hear mainly the mid frequencies and this makes it easy to check the balance of this critical area.

34. Have A Plan

Try to hear (imagine) the sound you want to achieve in your head before you start pushing buttons and turning knobs. With so much choice available these days you will get lost if you don't know beforehand what you are looking for. This also applies to synth or effect presets.

35. Vox & Bass Check

Before you commit to your final mix, wait until the next day if possible, and then do a final check on your vocal level and bass level. Those are the two places you are most likely to have misjudged things after spending a day mixing.

36. Keep A Clear Head

Please stay away from drugs, even if you are surrounded by them in the studio. Trust me on this one. Be the person whose judgement is not impaired.

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DPA 2017

Shotgun Microphone

DPA's new directional mic takes aim at an untapped mid-level market in location recording kit.

JG HARDING

I've used DPA mics for many years — my workhorses for documentary production are two DPA 4060 omnidirectional clip-ons. They are robust, low in noise and have a wonderful frequency response. They are not by any means a low-budget option, but are what you would genuinely call an investment in quality.

However, lavalier mics aren't the only option for this kind of work: many people like to use 'shotgun mics', or highly directional microphones, to capture dialogue when filming. DPA's new 2017 shotgun model is interesting in that it at first looks like a 'cut down' and much more affordable version of their existing 4017. This would be no bad thing at all, but in actual fact it also offers something a little different. The 2017 comes in just shy of half the price of the 4017, from which it inherits a lot of DPA DNA. The preamp and capsule design, though, are actually new, so it's more a child of the 4017 than it is a sibling, taking a different direction in a few interesting ways. Gone are the low-cut and high-boost switches, the modular preamp options and the lighter aluminium body, replaced by a fixed HPF and a hefty and invincible-feeling brass tube. So DPA have stripped out things that would add extra cost but not necessarily be useful to everyone, while also adding some differentiation, mainly in terms of the ultra-robust and compact construction.

I'm not a huge fan of reeling off specifications, as they're best presented by the manufacturer, but suffice it to say that the 2017 maintains the same kind of low noise, low distortion and high SPL handling found in the 4017 and its varied preamps. Notably, though, the weight is up from 73g to 115g, due to the brass body. If you're running a particularly long day on the boom mic and every gram counts, it's worth knowing what you're working with, though the 2017 is still lighter than many other shotguns.

The length of the interference tube itself — the cylindrical part with lots of slots in it — is actually comparable to that of many much longer mics, but the rest of the circuitry has been packed into a smaller space at the rear: an impressive feat, which gives the mic a slighter appearance than you'd expect, and marks a very welcome trend in the design of shotgun microphones.

The box includes the DPA mic and its interesting twist-lock mic holder, a zip pouch and a small windshield. A huge variety of DPA accessories is available for adapting the mic to various tasks, and accessories from both Rycote and Bubblebee Industries are now officially recommended too, so there is no shortage of choice available.

Though technically a 'budget' mic in the DPA world, it still reportedly works between -40 and +40 degrees Celsius and at up to 90 percent humidity, ticking off the vast majority of production environments.

Au Naturel

Along with a select few premium German mic manufacturers, I associate DPA with meticulous acoustic research and the desire to provide the most neutral frequency response possible. The 2017 does not deviate from this recipe, and as expected, the frequency response is pleasingly flat, with only a built-in high-lift and bass-cut coming into play.

DPA 2017
€1008

PROS
- Neutral frequency response.
- Ultra-robust brass construction.
- Low weight and small size.
- Low self-noise and high SPL handling.
- Includes foam windshield, stand clip and zip case.

CONS
- Fixed high-pass filter and high-lift may cause some to pick another product for more varied use cases.

SUMMARY
The DPA 2017 is positioned uniquely in the market. With very few sacrifices DPA have managed to significantly reduce the cost of a top-class shotgun mic, while adding a robust brass body. Only the fixed filter may give some pause for thought, but regardless, it's an excellent product that demands a response from competitors.

The 2017 is considerably shorter than most shotgun mics.

The low cut in the 2017 is fixed and permanent, a relatively steep third-order low-cut filter at 60Hz. For me this is a good choice of frequency, as I'm often frustrated by excessively high low-cut filter frequencies on preamps and mics. They often lead to me running 'open' while recording and leaving any filtering until the post-production stage. I'm also a fan of steep filters in this application, so the fixed slope was not an issue for me. It is something to bear in mind, though, if you're using the mic in a capacity other than location sound, such as recording an orchestra section or similar.

As with all directional microphones, working at 60cm or closer incurs a proximity boost in the low frequencies. This effect is useful for dialogue, and can be a help with close-up shots in production, as well as in music recording.

The 2017 also has a small fixed lift of 2dB at 15kHz. This is a useful feature in a shotgun mic, which will often be placed in protective or wind-blocking equipment that will naturally attenuate the high end. Like the low cut, this cannot be defeated. It's very gentle in nature, however, and is not the big 'vocal presence boost' found in some studio vocal mics, for example.

Although the 2017 is primarily aimed at location and film sound, there's no reason not to use a mic with very low noise, high pressure-handling capabilities and close-to-flat response for music recording. Individual small string instruments, for example, benefit from such a mic when recording an ensemble, and a hi-hat mic can often benefit from extra directionality and airiness. As a result the 2017 is not restricted exclusively to use on location.

Pickup Pattern

One very important aspect of a shotgun mic is its pickup pattern, and another is the linearity of the off-axis response. The off-axis attenuation on the 2017 is nice and tight: 10dB at 90 degrees and up to 20dB at the rear. The working principle of a shotgun mic is based on phase cancellation, the simplified explanation being that the long, vented tube itself acts as an acoustic filter. The longer the tube, the more sound from the sides is rejected by being bounced around in the tube and 'interfered' with. Obviously in reality this, like many acoustic principles, is vastly complex in nature. It's not currently possible to achieve absolutely perfect cancellation at all frequencies purely with acoustic engineering, but the more advanced computer modelling techniques become, the closer microphone designers come to creating the ideal system. The off-axis response of the 2017 has clearly benefited from such modelling in acoustic design, and is extremely neutral. I can't see any situation in which distracting coloration is going to arise in practice. We really are spoiled these days by the mics on offer!

"DPA have brought the price of exceptional-quality capture down significantly..."

On Test

In use, the 2017 was exceptionally resistant to handling noise, displayed no excessive sibilance, and had no issues with plosives in normal use. I was also unable to coax it into picking up RF interference, which is a good thing.

The noise floor always remained appreciably low even when recording at a distance, and as with most high-end mics, my location preamp showed its flaws before the mic did. No recording felt at all coloured or untrue, so I'm happy to report that there's not much to report! This kind of mic is designed to capture a true sound with a minimum of fuss, and that's what it does with aplomb.

That said, the 2017 is a more disruptive product than you might at first think. This kind of quality and pricing combination demands a response from DPA's competitors, as the mic currently occupies its own market position. By making very careful feature cuts, DPA have brought the price of exceptional-quality capture down significantly, and they've given the mic a rock-solid body to boot. If you don't care about the lack of switchable frequency adjustment or preamp options and don't mind the slightly higher weight compared with the 4017, it's hard not to recommend saving the money and getting a spectacular mic for a great price. Plug it in and enjoy it!

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ALTERNATIVES

The obvious alternative is the **Sennheiser MKH 416**, which has been a location-recording stalwart for many years. Originally released in the mid-1970s, the 416 has a very distinctive 'forward' character, with a 5dB bump centred roughly around 8kHz. Listening to recordings made with the 416 tends to reveal a high-midrange focus to the tone, and emphasised sibilants. It comes in at a very similar price to the 2017, but is an older design by many decades, plus it's quite a bit heavier. It is, though, a mic that pretty much everyone is used to. It's a bit like using an SM57 on snare: you know it's not a neutral sound, but you may well like how it sounds.

Some people recording on location prefer to hand off sound that is somewhat 'baked in' in order to minimise possible issues in post, while others aim for true neutrality. For the latter, the **Sennheiser MKH 8060** provides a similar neutral tone and directionality to the 2017 but at a markedly higher cost. The **Schoeps CMIT** series are also modern designs with commendably neutral performance but again at a higher price, and likewise the **Neumann KMR 81i**. All of these models have onboard filter switches. Below the £500 price mark there is a vast number of options, emphasising the 2017's quite unusual price point, sandwiched between budget-conscious and high end.

Korg OpSix SE & Wavestate SE

Synthesizers

Korg's Wavestate and OpSix are back, and they're bigger than ever...

GORDON REID

I got quite excited when I first heard about the Korg Wavestate because it promised to improve on the original and, in my view, wonderful Wavestation that I still use to this day. But when it appeared, it was a much smaller synth than I had expected, echoing the look and feel of Korg's Minilogue. With its pitch-bend and modulation wheels situated behind its parsimonious 37-note keyboard, no aftertouch and no control inputs other than for a sustain pedal, it was clearly aimed at the home studio rather than live performance, even though its sound generator was superior to that of its predecessor. I must admit that I was baffled by Korg's choice of hardware.

A few weeks later, I was equally excited by another Korg announcement. Tucked away behind a protective glass panel at NAMM 2020, it was called the OpSix and there was no doubting its lineage. Its name was obviously derived from 6-op FM synthesis and, with its metal case, five-octave keyboard and FM algorithms painted on its control panel, it was clearly a DX7 for the 21st Century. Offering subtractive, additive, virtual analogue and FM synthesis, the OpSix promised to be a mighty fine instrument. Then it arrived. It sounded great but, far from the original concept, it looked and felt much like the Wavestate.

When I reviewed them, I treated both the Wavestate and OpSix as desktop modules, playing them from much wider synths that generated aftertouch and had lots of control inputs. This made it possible to get the best from both of them. To be fair, Korg sell a lot of their smaller products (which suggests that I know somewhat less than I sometimes think) but I nonetheless concluded that synth engines of this depth and power deserved more professional, performance-oriented hardware. After all, a small synth satisfies only those who need a small synth whereas, unless space is very limited, a larger one satisfies everyone.

More Is More

Now, four years later, we appear to have the synths that the Wavestate and OpSix always should have been. Called the Wavestate SE and the OpSix SE, these are built upon metal chassis and boast 61-note velocity-, release velocity- and pressure-sensitive keybeds similar to that used on the Kronos 61. This gives them a much more robust and professional feel and, despite Korg retaining the dubious fashion statement of protruding Allen bolts to secure their upper panels, I think that they look great. What's more, they arrive in their own cases. These are solid and attractive, and will no doubt survive years of transportation in the back of your car. Having said that, I wouldn't trust them to

Korg OpSix SE & Wavestate SE

£1666 & £1899

PROS

- The Wavestate and OpSix synth engines deserved, nay... demanded these upgrades.
- Both synths create sounds that you'll obtain from nothing else.
- They're now robust and look much nicer than before.

CONS

- They would benefit from larger screens.
- They have no expression pedal inputs.
- External power supplies. Argh!

SUMMARY

The superb Wavestate and OpSix synth engines are now encapsulated in hardware that's more in keeping with their capabilities.

a tired and hungry road crew at 2 in the morning. If you're going on tour — and especially if you're travelling internationally — you'll still need pro cases.

This is all good stuff, so I'll now describe all of the improvements within their sound engines. Umm... I've finished. The architecture of the Wavestate SE is the same as that of the Wavestate and the recently announced Wavestate II, and that of the OpSix SE is identical with that of the original OpSix. The only significant difference, no doubt due to an iteration or two of Moore's Law, is that the Wavestate SE's polyphony is now 120 notes as opposed to the 64 of the original and the 96 of the Wavestate II, while that of the OpSix SE is 80 notes compared with the 32 of the original. Don't misunderstand me; these are valuable improvements, but they do not make new sounds available.

Are there any other differences? Not many. In addition to a handful of new Programs and Performances, the Wavestate SE offers 4GB of space for user samples that wasn't available when the first Wavestate was released. But, since this was introduced as a software upgrade in September 2021, I don't think that it counts. There are even fewer changes between the OpSix and the OpSix SE — ie. none — so it should come as no surprise that the Wavestate editor is compatible with all three models, and that the OpSix librarian is compatible with both the original and the SE.

This means that we now have three models of each synthesizer: a basic version with a plastic case, a 37-note keyboard, velocity sensitivity, and performance wheels located behind the keyboard; a standard SE with a metal chassis, a 61-note keyboard that adds aftertouch, performance wheels where they should be, and a case; and a sexy platinum SE that's otherwise identical with the standard one.

So, should you be interested in the new models, even if you weren't turned on by the smaller versions? I think that you should. Both of the originals were (and still are) powerful and fine-sounding synthesizers, but they were heavily constrained by their physical designs. The SEs make their power accessible to players who can take advantage of their enhanced keyboard widths and performance capabilities, so you may find that they're useful in exactly the way that the originals are not. Nonetheless, I have three reservations. The first is a common gripe... the use of external power supplies on performance-oriented keyboards. For me, this kinda shoots the whole concept in the foot. The second is the lack of controller inputs. With no way to connect an expression pedal or two, I still might find myself using a second keyboard to obtain the sounds and performances that I need. The third is the screen size. While a small OLED was suitable for the smaller models, a bigger screen such as that used in the Korg Krome series would be justified for the larger ones, as it would make programming both synths quicker and easier. I realise that this would have entailed software development as well as a more significant hardware redesign, but I think that it would have been worth it.

Finally, we come to the thorny issue of pricing. The short version is that you're paying really quite a lot more for increased polyphony and some purely physical changes. Can this be justified? I suppose that it depends upon what you want to do with them. If you're happy to use a Wavestate or OpSix as an expander, play narrow chords on them, use them to replay arpeggios and sequences, or even to use them as 37-note monosynths, the additional cost of the SEs might seem preposterous. But if you're a gigging musician for whom a more robust instrument, a 61-note keyboard and conventionally positioned performance wheels represent the bare minimum that you need, the smaller models are useless, no matter how affordable they are and how good they sound. Consequently, the Wavestate SE and OpSix SE might turn unusable synths into usable ones.

Conclusions

When I reviewed the original Wavestate and OpSix, I suggested that they should have been 61-note synths with aftertouch, lots of expression pedal inputs, and bigger screens. Well... we're halfway there, and you'll have to decide for yourself whether this particular glass is half full or half empty. But however you view them, the SEs are significant improvements over the earlier models. If money were no object, I might be tempted to snap up both of them. Caveats notwithstanding, they're amazing synthesizers that look great and can sound fabulous. ■■■

£ OpSix SE £1666, Wavestate SE £1899. Prices include VAT

W www.korg.com

The Rear Panels

Since these are repackaged versions, their recessed rear panels are identical with those of the originals and of each other. This means that the audio I/O is again limited to just quarter-inch L-R audio outputs and a quarter-inch stereo headphones output. There's only one pedal input (sustain) and only MIDI in and out (not thru) are provided, via 5-pin sockets. MIDI is also transmitted and received via USB but, as with every other digital synth or workstation with this omission, I'm disappointed to find that the USB doesn't carry audio. The final hole is a barrel socket for the 12V DC external power supply. You know what I think about external PSUs for performance synths, especially 2.5A units that you may be unable to replace at the local electrical store, so I won't rant again. I'll just say 'argh!'

SR Studio Classic Microphone Preamp

Two-channel Microphone Preamplifier

We try one of the keenest-priced 1073 clones currently available.

MATT HOUGHTON

SR Studio, or Stage Right Studio, is a sub-brand of electronics retailers Monoprice and under review here is what they describe as a "two-channel 1073-style microphone preamp". Monoprice's range in Europe isn't as extensive as in the USA and at the time of writing this particular device is available only in North America, but Monoprice tell me that if it sells well they may well roll it out in Europe too.

As I'm based in the UK, the review unit had to endure a transatlantic trip and, in its own cardboard box and suitable outer packaging, it arrived in good condition. Inside the box, along with the unit, were a pair of rack ears and the screws necessary to attach them, a transformer-based 60Hz, 120V AC to 24V DC 1.75A in-line 'wart' power supply with two-pin US mains plug, and a brief but helpful user-guide leaflet. UK mains is a 50Hz, 240V AC supply, so for my tests I ran the preamp plugged into a suitable step-down transformer.

SR Studio Classic Microphone Preamp

$599

PROS

- Appealing sonic character.
- Astounding bang for buck.
- Plenty of gain available.

CONS

- Noises at extreme DI settings.

SUMMARY

Under their SR Studio brand, Monoprice have delivered a good-sounding 1073 clone for a vanishingly low price.

Overview

Neve's 1073 started life half a century ago as a console preamp and EQ, and has earned a reputation for a flattering 'larger than life' character. This can be attributed in part to its Class-A transistorised circuitry generally, and partly to the specific effect of transformers in the signal path. The preamp stage has always had a transformer-balanced input and in the consoles, of course, the signal later flowed through output transformers. Most standalone preamps based on this design, then, have both input and output transformers. Some also incorporate the EQ section, and others provide an insert point for the EQ whereas some omit the EQ stage altogether.

SR Studio's asking price puts this preamp at under $300 per channel, making it one of the most aggressively priced 1073-derived preamps currently available. This is achieved through manufacture in Taiwan using 'no name' components, but while it has clearly been built to a price there are fewer corners cut than I'd anticipated. It remains faithful to the 1073 circuit, with discrete components in the signal path and transformer balanced inputs and outputs, and has an insert point for the EQ; there's no companion EQ in the range but you can hook up other 1073-style EQs (in tests, it played nicely with a GAP EQ73). The main gain controls are switched, so it's easy to match the channels for stereo mic arrays, and unlike on some clones there's an input impedance switch to cater for vintage mics designed with a low preamp input impedance in mind, but you can also use this to try and coax different 'voicings' from passive dynamic mics.

I was pleased to see a front-panel power on/off button (I hate it when rack gear can only be switched on/off at the back!) but on the review unit the power status LED didn't light. The remaining controls are laid out identically for each channel. A TS socket for the high-impedance instrument input is active when the adjacent DI button is depressed. When it isn't, a rear-panel combi input takes precedence, in which case the channel operates in mic or line mode, as set using a toggle switch above the instrument input. The remaining three buttons toggle the input impedance, 48V phantom power (which, incidentally, remains present on the combi connector's XLR and jack sockets even with the preamp in line mode) and the signal polarity. The blue, winged gain knob operates an 11-position switch, and has gain markings from 20 to 80 dB, the first two steps being 10dB each, and the others 5dB. The adjacent output knob comes after the EQ insert point and turns a continuous pot, so allows more precise control over the level. Finally, the top red LED of a four-LED output-level meter lights to warn that clipping is imminent.

Around the back, the XLR combi input connector can accept mic- or

line-level sources, as discussed, and the balanced outputs are paralleled on XLRs and TRS jacks to make it easy to plumb into your studio. The unbalanced insert send/return is a single TRS jack, and it's intended for use with a standalone unbalanced 1073 EQ section, though with the right cables you can safely hook up other gear. If an EQ is connected, it's always in the signal path — there's no insert bypass facility. On the inside, everything bar the output transformers sits on four PCBs. Two at the rear cater for the channel I/O, with one also receiving and distributing power. The other two are identical, carrying the shielded input transformer, preamp circuitry and controls for one channel each.

In Use

There's potentially a lot of bang for your buck here, so the all-important question is how it sounds. I tried this preamp on a number of sources (vocals, acoustic, electric and bass guitar, kick drum and snare), using dynamic and capacitor mics and DI'ing guitar and bass. I also compared it with a couple of my 1073-inspired preamps, a BAE DMP and a Golden Age Pre-73 DLX. With both an AKG C414 B-ULS and a Heil Sound PR40 it worked very well on my own voice. It wasn't perhaps my favourite sound for delicate acoustic guitar parts, but to be fair that's something I'd say about the 1073 generally! DI'ed bass and guitar sounded rich and full. It was also possible to drive the gain harder and bring down the output level to compensate, to coax more character from the electronics. There was always plenty of gain available, and the sound was certainly a touch richer/fuller than that conveyed by, say, my RME or Audient interface preamps.

Largely, then, it's a good-sounding, flattering preamp that works well on a range of sources, but is there anything not to like? When driving it harder for character, particularly when playing DI bass, there were definitely differences compared with my BAE: a touch more noise (albeit not so high as to be problematic), and the nature of deliberate distortion wasn't quite so enticing. It wasn't unpleasing, though, and note that I'm comparing it with a device that's five times the price! A small quirk I should mention is that when the DI is switched in (with an instrument connected or not) and gain is set to 75 or 80 dB, then if you turn the output knob past about the 3 or 4 o'clock position, the noise floor rises significantly and it sounds as though something's being driven into self-oscillation. These are extreme settings that I'd never want to use in the real world, so it won't really be a problem in practice, but it didn't happen with my other 1073-ish preamps and it would be remiss of me not to mention it.

Worth It?

The character of a mic preamp is a pretty subtle thing in the grand scheme of things, compared with, say, your choice of instrument, mic and other processors, and it's fair to say that the preamps built into most audio interfaces today are perfectly capable of good, clean recordings. But this one does offfer a different sound character and it can be controlled by juggling the gain and trim controls, so it will no doubt appeal to many project-studio owners. It also offers a huge amount more gain than most interface preamps, which some will find useful; has switched gain controls, which is great if you want to match two channels or recall settings; and puts some useful facilities within reach on the front panel.

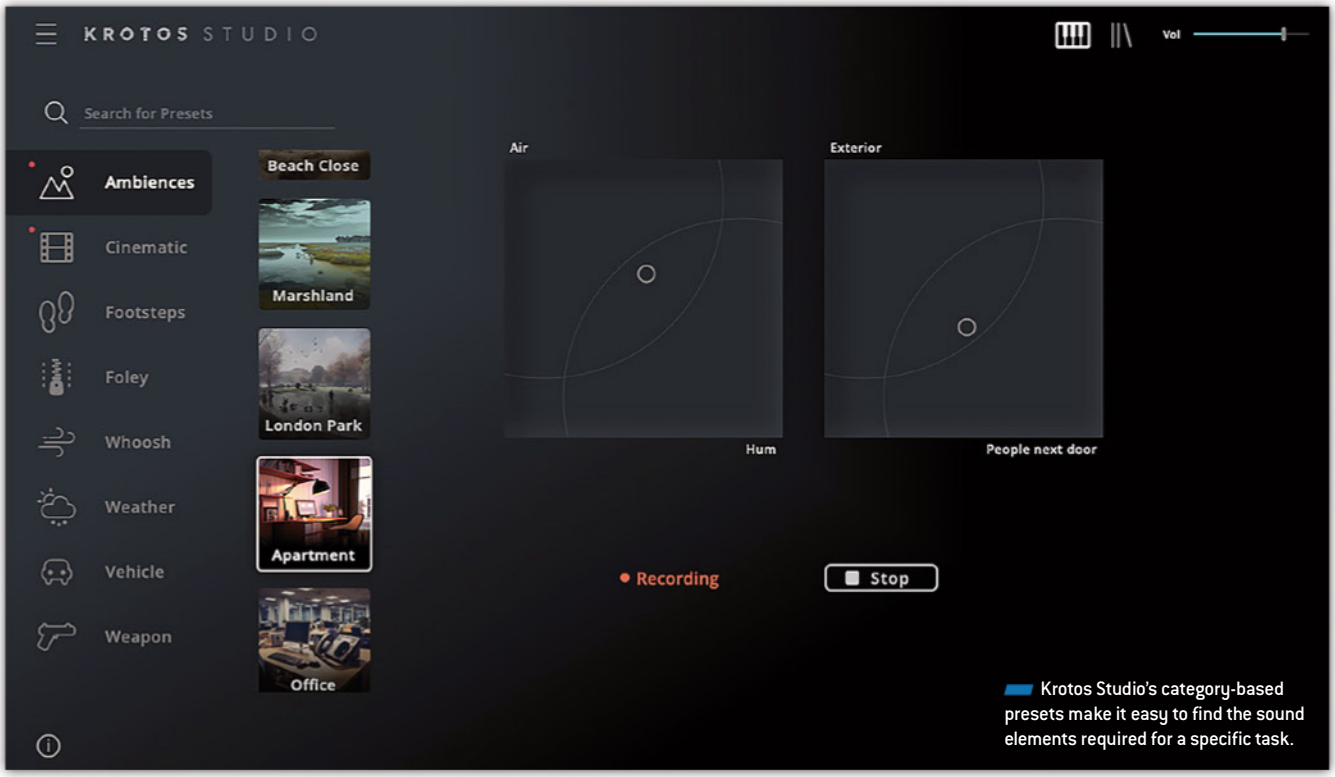
Yes, it's possible to tell the difference between a clone built to this price and a vintage Neve or no-holds-barred replica, but it has that broad family character. The biggest sonic differences will no doubt be down to component choice, particularly the transformers — there are four here and if you replaced them with the most desirable transformers used in 1073 replicas that'd set you back over $200, which is a good chunk of the asking price! I may not personally be about to throw away my favourite preamps for this one, but the SR Studio 1073-style preamp is way more appealing than it has any right to be at this price! ■■■

An insert point allows 1073-style EQs to be connected to either channel or both.

ALTERNATIVES

There are lots of standalone 1073-inspired preamps, but the only one I've used that is more affordable on a price-per-channel basis is **Golden Age Project**'s **Pre-73 Jr**.

£ $599 excluding taxes and shipping.
W www.monoprice.com



Krotos Studio's category-based presets make it easy to find the sound elements required for a specific task.

Krotos Studio

Sound Design Software

Krotos Studio makes custom sound design accessible to almost anyone.

JOHN WALDEN

Sounds created with Krotos Audio's products have found their way into major films and AAA video games so, when they release a sound-design tool that claims to be easy to use and is accessibly priced, it's likely to get budding sound designers paying attention. Whether you are new to sound design and need a simple-to-use creation process, a solo video or game creator doing their own sound design (as well as everything else), or an experienced sound designer needing a tool to get the job done fast when the deadlines are stacking up, is Krotos Studio a potential solution? Let's explore...

In The Studio

From a technical perspective, Krotos Studio can run as a standalone application or as a VST, AU or AAX plug-in. For the purposes of this review, I mostly used the VST plug-in within Cubase Pro 12 but both it and the standalone version ran smoothly on my test system. The product installs in two parts; the base application/plug-in and the sound content (currently around 17GB; this can be located on any drive). As well as the core Factory Content that gets you started, subscribers also receive access to new sound content each month.

The package is available on a subscription-only basis, with either annual or monthly plans. Yes, for those just starting out, or working on a hobby basis, perhaps subscription-based tools are not always their first choice, but for start-up businesses, being able to set a fixed monthly budget for things like stock video footage, stock music, plug-in suites — and now sound-design tools — is an attractive and sensible way to keep budgets under control. Krotos Audio do also offer a free version with a limited amount of content so you can try the software before making a commitment. As made very clear on the website and in the documentation, an active Internet connection is required, presumably to confirm the status of your subscription when in use.

Perhaps the other obvious technical feature to note is that the sound creation is in stereo and, currently at least, does not provide support for surround-sound formats. However, in suitable hosts, you can use multiple stereo output channels (different sound elements are automatically routed to different output channels) so you could certainly generate audio that might then be used in a surround context.

Welcome To My Pad

Krotos Studio can be loaded into your host/DAW project like any other virtual instrument. You are then presented with a very intuitive UI. On the left, you get a list of sound-design categories — Ambiences, Cinematic, Footsteps, Foley, Whoosh, etc — and, if you click on one of these, a set of sound presets appears for that category, each with a visual icon and a label. Some of the categories contain lots of presets. For example, at the time of the review, the Ambiences category contained over 40 different presets such as city environments, forest, suburbs, building sites, the seaside, parks, offices, bathrooms, restaurants, airports, deserts

and, just for good measure, a zombie apocalypse. Other categories currently have fewer options so, for example, the Vehicle category has just three entries. However, it's additional presets that form the basis of the new content added every month. For example, the September 2023 download (which became available as I was working on this review) added 13 new presets, spread across a number of the core categories, with a total of 1.7GB of new content, all at 48kHz/24-bit.

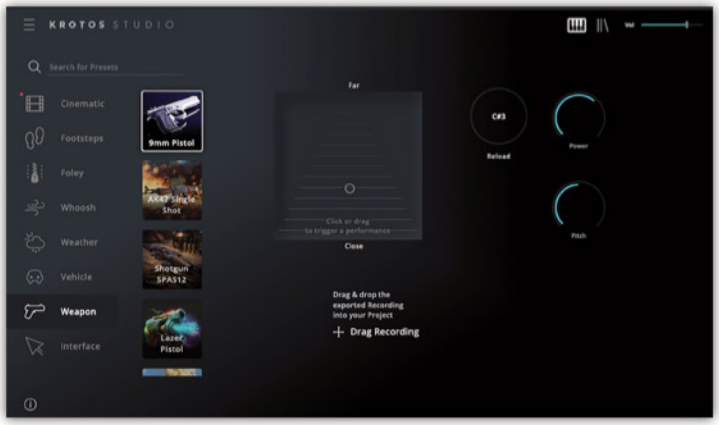
Once you have picked a preset, the right-hand side of the UI then displays a combination of X/Y pads, trigger keys and circular volume sliders that allow you to 'play' the preset's underlying sounds. This can be done via the mouse or, if you have a suitable MIDI controller connected, via a combination of rotary controllers (for the X/Y pads) or MIDI notes (for the trigger keys). These mean you can also record and/or automate your sound-design performance but, by default, and rather neatly, Krotos Studio records audio in the background as you tweak the controls to design how you want the sounds to evolve or the timing of spot sounds to occur. When done, you simply click on the Stop button and a 'Drag Recording' prompt appears. You can then drag and drop the performance you just created as an audio file onto a suitable audio track to be positioned on the timeline and, if required, subjected to any further editing/processing. It really is a very simple — and very speedy — process.

The nature of the control combinations varies depending upon the preset type. For example, for many of the ambience presets you get a combination of two or three X/Y pads, each of which offers different sonic elements on their axes; you simply blend the various elements as required. Taking the Forest preset as an example, the two X/Y pads provide birds/wind and stream/animals; you can simply dial in, and change, the combination of these sounds to suit your needs. In the Whoosh category presets, one X/Y pad provides what is termed a 'peak pad', where moving across the X axis allows you to build the intensity of the sound, peak in the centre, and then provide a drop in intensity. You can therefore control the overall speed of your 'whoosh' effect to sync with your visuals. And, in the Weapon category, the Y axis in many of the presets lets you vary the distance of the sound (for example, a gunshot or explosion) relative to the listener's position (near to far).

Inspiring Sound

With the 'ease of use' box firmly ticked, what about the sounds themselves? Based upon the current content, the quality of the sounds is very good. For example, the ambiences (real or imagined) or the weather-based presets are very effective and, with suitable use of the X/Y pads, you could easily create extended beds from these sources without a sense of repetition creeping in. The range of Foley sounds is also good, with presets for various types of doors, switches, tools, keyboards, paper, lockers, water and a few 'horror' options... in case you need to transition between 'bone snap' and 'brain crush', for example! I suspect the recording process for some of the underlying sound sources will have been fun.

The vehicle category perhaps felt a little underpopulated at this stage, with just three presets. Yes, they could be used in a wide range of contexts, and easily mixed with presets from other categories for additional variety. Hopefully, the monthly additions will add to the existing content in this particular category. However, the sound designer's favourite — the Footsteps category — is very well populated, while the Weapon category has a good combination of small arms and large explosions, all with plenty of control to add variability. There is also a neat Interface category with various computer bleeps, bloops and notification sounds.



Each preset offers a specific control set, as seen here for the '9mm Pistol' preset within the Weapon category.

Krotos Studio

From £14/month

PROS

- Custom sound design made very easy.
- Subscription model lets you try without a significant commitment.

CONS

- While the content is expanded each month, more experienced sound designers might find the current selection limited.
- Not everyone loves subscription-based tools.

SUMMARY

Krotos Studio makes custom sound design accessible to almost anyone. An interesting concept that ought to appeal to independent media creators.

Conclusions

So, the quality of the existing sounds is good, and the UI makes the design process — customising the supplied sound content to suit your visuals — accessible to even the most novice of users. You can, of course, add additional variety to the sounds by further processing them within your host (via EQ or reverb/delay, for example), but to keep users coming back, and to allow them to see Krotos Studio as a primary sound-design platform for larger projects, I suspect that fresh monthly content is likely to be critical.

OK, so maybe you wouldn't expect to fully-furnish a Hollywood action blockbuster with sound effects based solely on Krotos Studio, but it could certainly do good service for lots of routine sound elements or in projects of a more modest scale. What's here is certainly a very encouraging starting point and, for media creators looking for an accessible source of sound-design elements, a single-month subscription would easily let you evaluate it for yourself. Whether it's the sound of gentle rain, an urban soundscape, a sci-fi transition, or brains being crushed(!), sound design really doesn't get much easier than Krotos Studio.

£ £14.99 per month/£119.88 per year. Prices include VAT.
W www.krotosaudio.com

Yamaha CK61 & CK88

Stage Keyboards

In terms of pure bang for your buck it doesn't get much better than Yamaha's CK keyboards.

ROBIN BIGWOOD

The CK61 and CK88 on test here represent a completely new range from Yamaha. They join a stage keyboard line-up that is still spearheaded by the more expensive CP and YC series I reviewed in *SOS* back in September 2020, but seem to have acres of overlap with them. What are Yamaha thinking, and what compromises have been made?

Two Become One

I'll start by thanking Yamaha: the CK61 and CK88 are very easy to review and describe as a pair, with identical capabilities, and only physical differences.

The CK61 has a five-octave velocity-sensitive keyboard with Yamaha's FSB (Future System Basic) action, which apparently was developed for their Electone organ range. It has a slightly narrow 160mm octave span (as does the YC61), compact 14cm-deep key tops, and a touch which offers strong springy resistance in the first few millimetres of travel, before the key really falls. It makes for a reasonable piano-playing experience from an otherwise fast synth-like action.

Meanwhile, the CK88 has a piano-like hammer action, using the Yamaha GHS (Graded Hammer Standard) keybed. That represents the first rung of Yamaha's piano action ladder, but it's perfectly nice for typical stage and studio use. There's a normal 165mm octave span, spacious 15cm-deep white key tops, sculpted black key fronts, and a precise, modern-feeling 10mm key dip. No 'escapement' resistance of Yamaha's high-end Clavinovas, but certainly a supportive, swinging release. It's mechanically quiet too, especially compared to something like a Fatar TP40, with little more than a dull thud at the bottom of the downstroke. As on the CK61, there's no aftertouch sensitivity.

Then the only other difference is in the provision of mount points for a YMR-03 music rest: the CK88 has them (though the rest is not provided as standard) and the CK61 doesn't.

Common Knowledge

As to what the CKs have in common: well, how long have you got? Despite being at the affordable end of the market, these keyboards have feature sets that are surprisingly extensive. So much so in fact that there's no way I can cover everything here, though I'll try my best to pick out all the really interesting and pertinent points.

The headline features start with the provision of a do-it-all soundset: tonewheel and transistor organs (with physical drawbars), some top-class acoustic and electric pianos, and (to use Yamaha's own front-panel button groupings) brass and wind, guitar and bass, strings, synth pads and leads, chromatic percussion, and a few 'others'. Approaching 400 preset sounds in total, which is more than the current YC or CP series can offer put together, even after a series of firmware updates that have extended their abilities.

You employ these presets in a three-part multitimbral architecture, which proves to be a doddle to work with thanks to dedicated level faders and part on/off and selection keys just to the left of the central 128 x 64-pixel white-on-dark LCD display. Any type of sound can be loaded to any part, so it's quite possible to have three organs or three pianos on the go at once, or to start layering up synths alongside more bread-and-butter split-keyboard combos. What's more, each part gets its own independent effects chain. That includes a drive section (which also undertakes Leslie speaker duties), plus two multi-effects engines. There's a further master-level delay, reverb and three-band EQ.

Entire three-part setups, encompassing sound selection, flexible keyboard splits and layers, transpositions, effects and additional settings can be saved to a kind of 'multi' memory called a Live Set. There's room for 160 of these, directly accessible from eight front-panel buttons across 20 'pages' from which they load instantly. They can further be liberated to a plugged-in USB thumb drive, either individually, in page-size chunks, or all in one go. There's also a global option to move forward and back through Live Sets one at a time using an appropriately configured foot pedal.

And talking of pedals, real-time control provision on the whole is rather good. Two pedals can be attached — switch or expression types in either socket — and their function determined on a Live Set (rather than global) basis. They're by no means limited to sustain and level duties: either or both can drive virtually any parameter that has a hardware control, including the volume of individual parts or the whole instrument.

The front panel manages to be busy but clear. The drawbars are very serviceable, despite their only 21mm travel, and the action of all buttons and knobs is very nice indeed. No wobbles or flakiness anywhere. And it's worth noting how many single-function buttons there are too, both for real-time sound control (mono keyboard mode and unison) and functional operations (keyboard split and Live Set store).

All buttons that can benefit from it are capable of being backlit, to show on/off status: they can even do a variety of (user selectable) colours, which helps confirm which part is selected. As an aside it's worth noting that the backlights seem »

Yamaha CK61 & CK88

£749 & £1149

PROS

- Flagship pianos, Hammond and transistor organs, and a big general-purpose sound set, with plenty of real-time control.
- Flexible three-part multitimbral architecture.
- Very easy to use.
- Brilliant value for money.

CONS

- Keyboard actions aren't bad, but there are better.
- Choice of acoustic and electric pianos restricted, compared to more expensive competitors.

SUMMARY

Versatile, fine-sounding, budget-friendly stage keyboards that can stand with much more expensive opposition, the CK61 and CK88 are stonking bargains.

One More Thing

Actually who am I kidding? It's lots of things. Just when you thought the CK feature set was already punching above its weight, there are a handful of unusual, 'edge-case', but possibly very useful features to discover.

First, the CKs have built-in speakers: a pair of oval 12 x 6cm drivers, rated at 6W, mounted underneath the chassis and firing downwards. They lack both oomph and bandwidth, but they might prove to be tremendously useful every now and then. Inserted headphones silence them, and if you'd rather they didn't exist at all a rear-panel switch turns them off.

Going hand in hand is a battery power option. Eight AA batteries will power a CK for between about three and five hours, and the LCD screen shows charge status. Power comes otherwise from the supplied 12V AC adaptor.

Then there's the 'A/D input'. It's a pair of quarter-inch sockets and associated rear-panel gain pot that lets you plug an external line-level device, or indeed a microphone (which needs to be self-powered, as there's no phantom or other power delivery here), and bring audio into your CK. Surprisingly, and impressively, there's a dedicated signal path for this input that includes level control, two multi-algorithm effect processors (the equal of those provided for each instrument part, though without any hardware controls), plus a three-band EQ and a gate. There's even a front-panel button that takes you directly to all the associated settings, which can be saved into a Live Set memory. There isn't a vocoder, nor any vocal harmonisation options, but still.

There's another way of getting external audio to the outputs of your CK: a stereo Bluetooth audio input, absolutely ideal for backing tracks. You can also designate the highest or lowest key to trigger the playback of a 44.1kHz 16-bit stereo WAV file on an inserted USB thumb drive. It can be a different file for each Live Set.

Then, attach the CK to a computer or iPad/iPhone via USB and it's seen as both a MIDI device and an audio interface. Sample rate for the latter is a fixed 44.1kHz, but it's capable of reasonably low latency: on my Mac a 64-sample sample buffer in PreSonus Studio One gave a reported input latency of about 6ms and output of 4.5ms. The stereo USB input to your computer carries part sounds plus signals coming in via the A/D input and Bluetooth. In the other direction, stereo playback can be monitored via the CK's internal speakers, headphones, or attached monitors. You might not record your next orchestral session with just a CK and an iPad, but you could get a heck of a lot done in your bedroom.

The CKs' back-panel connections are identical, featuring a 12V power input, quarter-inch audio I/O and foot pedal inputs, MIDI I/O, a pair of USB ports and a switch for the built-in speakers.

» to employ a 'field-sequential' design, like some digital camera viewfinders, whereby some static colours are generated by rapidly strobing others. Glance quickly left or right while using your CK and you might see flickering or colour 'tears', which can be a bit disconcerting at first. You get used to it.

I'll quickly mention two other interesting abilities, before we dig a bit more into the detail of the sounds on offer. First, any (or all) of the three multitimbral parts can be exclusively driven via an external MIDI keyboard. This gives the possibility of using the CK61, say, with an 88-note controller keyboard, for that 'best of both worlds' setup where pianos are played from a hammer action, and synth or organ sounds from the built-in lighter-action keyboard. A parallel scenario can easily be imagined for supplementing the CK88 with a synth-style or waterfall organ keyboard. Or, it's a great way to incorporate a MIDI organ pedalboard. The way this is done is simple and clear, and a Live Set's part list will show 'EX' next to any sound set being driven externally.

Second, a Master Keyboard mode, which happily runs alongside internal sounds, lets you define four keyboard zones that can transmit MIDI messages to external gear, with their own channels and transpositions. On recalling a Live Set where this mode is active, the CKs can even send out MIDI bank and Program Change messages to coordinate external synths, and there are per-zone options to transmit (or not) messages from pitch and mod wheels, and the pedals. Master Keyboard mode is hidden in the menu system, but is easy enough to work with, and extends the CK's functionality in a bigger setup enormously.

Sound

All the facilities in the world aren't worth much if not backed up by some good sounds, but it turns out the CKs come with an impressively broad and high-quality sound set.

Let's start with the organs. Five different types use the drawbars and associated percussion and vibrato/chorus controls. They are emulations, to differing degrees of historical accuracy, of a Hammond, Vox Continental, Farfisa Compact, Acetone (a 1960s Japanese transistor organ) and Yamaha's YC-45 from the early '70s. In general, these are excellent, and exude character. They're not without their quirks, though.

The Hammond does everything it should in terms of foldback of upper footages and percussion retriggering. The Leslie speaker sound, should you want it, is provided by one of two algorithms in the drive effect processor, Rotary A and Rotary B, and it's to this processor that the two dedicated Rotary buttons at far left of the panel (the equivalent of a 'half-moon' switch) are directly connected. Rotary A is a respectful, contained, light-leaning sound, though capable of generating some very convincing and likeable overdrive distortion when the Depth knob is turned up. Rotary B has much longer hair, so to speak. There are far higher levels of a thicker distortion (which, at its extreme, gets a bit unpleasant), greater stereo width, and this cabinet is simply louder. It has a slower Fast rotor speed too.

Now to the transistor organs. These are better, in my judgment, than those in the YC series, and there are more to choose from as well. Here the Vox is all bright square waves, the Farfisa fatter sawtooths, the Acetone a dirty pulse with a bit of sub-octave bleed, and the old YC model more or less a brighter Vox with less key-on 'tack'. They are still only approximations — there's no attempt to recreate a Vox's flute and reed tones, for example, nor its limited number of drawbars — but all can survive close scrutiny without needing to be smothered in effects. The percussion section is in play (and here, correctly, without that Hammond 'discharge'). Vibrato/chorus can be enabled too, to good effect, though I wonder if the manic wobble rate of the C1/2/3 modes is a bug waiting to be fixed. »

Synth Vs Stage

Stage keyboards always have to strike a balance between ease of use and flexibility, and their designers almost always opt not to expose the full synth architecture they undoubtedly have within. That's true here.

On the CKs you get to tweak sounds via dedicated front-panel knobs: filter cutoff and resonance for a low-pass filter, and attack and release time for an envelope generator. All work in a relative rather than absolute fashion: a nominal 'zero' point, representing a sound's default factory-programmed parameter value, is always at their 12 o'clock position.

You can do quite a bit with these four knobs, but their behaviour varies from sound to sound. Most pianos and organs can't be made brighter or to attack more quickly than they already do, but many synth pad and lead sounds can. All sounds can be darkened by reducing filter cutoff, but usually not completely silenced, and you have no control over filter slope. Creating slow, lingering pads by increasing attack and release times generally works nicely, but the scaling of values is surprising: releases get very long quite quickly, and at the max setting a typical pad sound will take well over a minute to disappear. Drone artists, fill your boots.

Allied with the part effects, and the multitimbral design, it's entirely feasible to conjure up exotic and complex timbres from straight-laced starting points. The Unison feature helps with this too, generating a simple (and adjustable) chorus, or super- and sub-octave stacks. The extent to which the modulation wheel affects pitch, filter cutoff or amp level, and at what rate, is also up for grabs.

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Auto Gain automatically sets your levels and Clip Safe keeps them in the sweet spot. Plus a huge bundle of software and plugins.

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» Non-drawbar organs include nine pipe organs, four electric-type church organs, an accordion and a musette. Most are very good, finding strength in more simple registrations than is common in stage keyboards. The accordions are superb too: a rarity.

The acoustic piano category runs to 13 sounds, but in fact six of those are different versions (mono, bright and warm-balanced) of the excellent, luxurious, contemporary-sounding CFX grand that has been Yamaha's flagship for years now. It remains a pleasure to play, and has neither discernible velocity switching nor any glaring transitions in timbre from note to note. Held decays have good complexity for several seconds, and are fine even beyond that. A neat S700 studio grand and a plummy-sounding 'Live CF3' have similarly good manners, and a U1 upright offers character in abundance.

The sound engine is not sophisticated enough to support sympathetic resonance, and sustain pedal bloom is generated by a Damper Resonance layer effect that will occupy one effect slot of the part the piano is loaded to, if you choose to use it. At default settings it sounds natural and convincing, but can become anything but, like a huge pulsating reverb, if you crank up the effect knobs. A Yamaha FC3A or other continuous-type sustain pedal will support half- and partial-pedalling for all acoustic piano sounds, and very nicely too.

Electric pianos are represented by two Rhodes models and a warm and bright Wurlitzer. Not a lavish selection then, but once again they're of excellent quality, with great velocity response, and some key-off noises. Pedal response is subtly accurate too, with partial damping possible, but not half-pedalling.

Transitional Period

The ability of a keyboard to load new sounds without interrupting any that are already sounding is highly prized in some circles. The CK61 and 88 don't promise they're able to do this, but often can, with certain provisos. Suffice to say, changing a part sound or loading a new Live Set while holding notes causes no nasty clicks, and almost never silence, and it's effects blocks switching on or off, or changing algorithm, that give the game away more than anything. This is an improvement on CP-series keyboards of old, and is impressive at the price.

Further electric piano sounds include two clavinets, perfectly fine, but (like on the CP and YC series) an octave lower than everything else. It's ironic that the six FM pianos exhibit velocity-driven lurches in timbre, and are almost certainly generated from samples rather than a true FM engine. In practice they work perfectly well, though.

As for all other sounds, there's not enough space here to go into much detail. To ruthlessly summarise though, there's both breadth of coverage and impressively high quality. Brass/wind sounds are a highlight, with sassy and highly playable trumpets, trombones and saxes in both solo and combo groupings. Acoustic guitars and basses are way better than you might expect them to be, expressive and convincing when played well. Amongst the squelchy synth basses some plain fundamental tones are very welcome, working by themselves and as ideal layer components.

The over 100 string, pad and choir sounds give masses of variation in timbre, style and scale, and often sound very classy indeed. I particularly liked the handful of string machines nestled amongst the Jupiters and Oberheims. Fifty-one synth lead sounds start out polyphonic, but can be made mono, with different flavours of portamento, and cover everything from typical '70s funk to more modern EDM.

What the CK series does not have is any drum kits. There's a good helping of bells and mallet instruments in 'Chr.Perc', and a smattering of folk and ethnic instruments (plus timpani) in 'Others', but you'll have to look elsewhere for your beats.

Effective

As I mentioned, each CK part gets three effect processors of its own. The drive section, it could be argued, is more about rotary speaker emulations than anything else. Overdrive and distortion here is fine, quite organic and complex, but the Comp(ressor) is for my money too polite: it works, and adds some weight, but

The front-panel controls are also the same, offering hands-on control over a number of commonly used parameters and some frankly adorable drawbar sliders. All together now, ahhh.

turned right up is the point at where I'd like it to be getting going.

The twin identical multi-effects engines are pretty impressive. I counted 36 different treatments to choose from, within those front-panel categories, and they sound very good. All the choruses and more psychedelic offshoots hit the mark, and there are further guitar amp sims and another Leslie, for distortion fetishists. Aside from several EQs are threshold- and pedal-driven wahs, static low- and high-pass resonant filters, and a sample-rate reducer for lo-fi effects. Three straightforward delays are offered (digital, analogue and ping-pong) and the reverbs (hall, room and reverse) are quite respectable, capable of natural-sounding and more synthetic results alike. The simple pairing of Depth and Rate knobs usually allows a wide variety of settings to be teased out.

Turning to the master-level effects, the reverb there is subtly different, with a plate offered instead of the reverse algorithm. There's also only one knob, fulfilling mix and other parameter duties at once, which works well enough but isn't ideal. The master delay's Depth and Time knobs let you do more, and a Tempo Delay algorithm lets you set time in bpm values, or via rhythmic taps of the Enter button.

Finally, the three-band EQ. Offering a boost or cut of 12dB in each band it's more spatula than scalpel, but is effective in what it does. There is a slight conflict of role, relating to the way it sometimes acts at a global level, and at others has settings recalled by a Live Set, but it just about makes sense in the end.

One small down side of the master effects is that individual instrument parts don't 'send' to them (which would allow you to have master reverb on one part and not another, for example) but they instead act on the entire mixed output of the keyboard parts. The presence of part-level effects stops this being a problem as such, but there's not the useful additional flexibility there might have been.

ConKlusion

There is a certain studied lack of flashiness about the CK61 and 88 that might make you assume they're deliberately feature-starved and probably not for consideration unless you're on a tight budget. The reality couldn't be more different. Having spent several weeks with the CKs I think they can hold their own alongside models twice or three times the price, by Yamaha, Korg, Nord and others.

It's true that some aspects of design are a way from state-of-the-art. For example, the keyboard actions here are not Yamaha's very best. Drawbars are short-throw and aren't a patch on the YC series' long, LED-backlit designs.

However, the GHS and FSB are not bad keyboard actions. Far from it — they're more than fit for purpose, and they're light years ahead of the clunky counterparts found on many cheaper MIDI controllers. Drawbars may be short, but they're nice to use and can be set accurately.

In most other ways, then, I struggle to pinpoint any major drawbacks. Yes, the CK's acoustic and piano line-up is much smaller than that on the CP and YC series, let alone a recent Nord, but the excellent CFX grand sample is there, alongside several likeable, responsive Rhodes and Wurlitzer sounds. I'll hazard a guess they're what most users will be more than happy to stick with and end up using most of the time. The Hammond sounds great, the transistor organs are buzzy and ballsy, and the two Leslie simulations, very different from each other, are equally useful and fine-sounding. The remainder of the sound set is stuffed with goodies that will only be bettered by computer-based sample libraries and a few big-money workstation synths.

The three-part multitimbral design, and the way it's been implemented in such a clear way, with dedicated controls and independent effects, is to my mind a big improvement over the YC and CP ranges. I prefer the effects implementation generally too. The knobs may be smaller, but as ever, size isn't everything. Nothing about the user interface feels complicated or arcane, and it was only out of a sense of duty I ever felt the need to consult the manual.

In the final reckoning I've nothing but praise for the CK61 and CK88. I'd buy one without hesitation if I were looking to invest in an honest all-round stage keyboard. They're a brilliant option if the budget is tight, but you shouldn't overlook them even if it isn't. They should fit right into diverse studio, educational and domestic environments too. In a period in which we all seem to be paying more for less, the Yamaha CK series bucks the trend, and then some.

£ CK61 £749, CK88 £1149. Prices include VAT.
W www.yamaha.com

SPHERE SOUND

The Edge, Joe O'Herlihy & Steve Lillywhite: Creating *U2:UV*

U2's residency at the Sphere in Las Vegas is revolutionising live music.

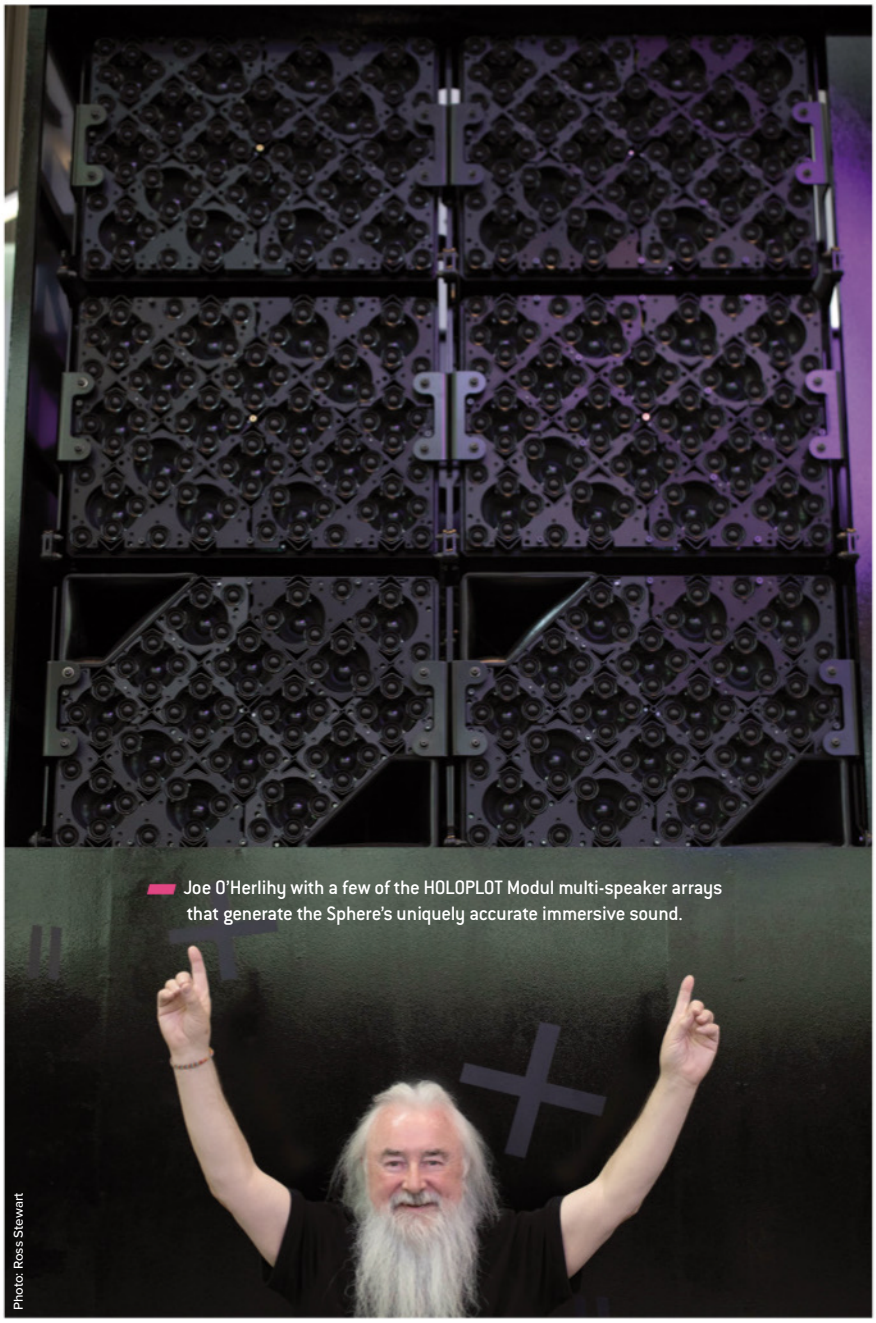
Photo: Rich Furey

TOM DOYLE

Since opening at the end of September, the Sphere in Las Vegas has been widely hailed as the next level in live entertainment, thanks in large part to U2's inaugural residency there. Titled *U2:UV* and based around the band's 1991 album *Achtung Baby*, it's currently set to run until February 2024.

The stats for the Sphere are staggering. Some 366 feet tall and 516 feet at its widest, it's the largest spherical structure in the world. Inside, it boasts the largest high-resolution (16k) LED screen on the planet, curving up, over and around the venue's capacity audience of 20,000. The exterior of the building, meanwhile, features a 580,000-square-foot 2k-resolution screen, which beams striking imagery — a yellow blob emoji, a moving eyeball, the face of U2's animated 'space baby' — out over the Vegas skyline.

Then there's the ground-breaking audio system, designed by German company HOLOPLOT and named Sphere Immersive Sound. Comprising 1534 fixed and 300 mobile units of their X1 Matrix Array speaker modules (both their two-way Modul 96s and three-way Modul 80-S variations, with 167,000 drivers in total), it is largely hidden behind the LED screen, and promises the clearest concert sound on the planet. Featuring HOLOPLOT's 3D Audio-Beamforming technology, which can project sound, on both vertical and horizontal axes, into very specific areas, even individual seats, and Wave Field Synthesis algorithms to maintain consistent quality of sound waves over distances, Sphere Immersive Sound represents a complete rethink of how live audio works.



Joe O'Herlihy with a few of the HOLOPLOT Modul multi-speaker arrays that generate the Sphere's uniquely accurate immersive sound.

Photo: Ross Stewart

Clarity

Both U2 guitarist The Edge and Joe O'Herlihy, the band's live sound man of 45 years (see box), reckon the Sphere system is a gamechanger. "We've never had this level of clarity," says the former. "So, we've never felt that the detail of what we're doing could be appreciated to this level. As much as it is a kind of throwdown, it's also an opportunity to really work very hard to deliver a level of dynamics and subtlety in the arrangements that we would have done in the past, but would have been kind of lost. If you try and play really quiet in a stadium, no one's going to hear you. The wind will blow it away. Whereas, in this case, you really can explore a different type of contrast between the loud and the soft parts in a song. You can play very quietly, and it will carry. So, it's an exciting option."

"This HOLOPLOT system is basically the future," O'Herlihy says. "Audio has been waiting for a development like this for such a long time. The sonic value for me is fantastic. The band have always been very much audio-oriented in the sense of: we want to give our audience the best possible perspective from an audio point of view."

SOS attended the opening night of the Sphere and can attest to its — and *U2:UV*'s — dazzling wonder. Audio-wise, the surround features in the production are actually used sparingly, which arguably gives them greater impact. There's the moment in 'The Fly' where The Edge's guitar solo descends from the roof. At the beginning of 'Where The Streets Have No Name', the organ intro creeps from the back of the dome over the audience's heads and towards the stage in time for the band to kick in. But, overall, U2 and O'Herlihy's use of the immersive

sound system emphasises clarity over novelty. “We don’t want to be involved in sort of gimmickry,” Edge stresses. “Sound is still the most important medium for any concert. But it’s not the technology that’s important: it’s what the sound delivers to the audience in terms of the emotional connection. We don’t want to get lost in the options in the technology. What we’re trying to do is just find a way to enhance the experience and make it as potent and powerful as possible.”

“We didn’t go overboard, as such, in what we could have done,” O’Herlihy adds. “I think we were very delicate in the application, in the sense of… we didn’t want to turn it into a movie. It’s a rock & roll gig, after all.”

One standout feature in the show is when the choral voices in the middle of ‘Beautiful Day’ fan out across the dome. “We have this sort of vocal moment,” says Edge. “We’ve put the voices in different parts of the building, to really feel like the audience are in the middle of the sound, rather than it coming from the stage in their direction.”

“So you have a really kind of atmospheric backing vocal section,” O’Herlihy says. “It’s a joy to be able to do that. We’ve gone from, you know, stumbling through our analogue world, into the digital domain, to catching up with the rest of the world in the context of audio. I mean, I’ve mixed in a particular way for years and years and years. But when I was introduced to this, I had to sort of refocus my head.”

Plotting

Plans for U2 to open the Sphere were hatched back in 2021, but were complicated by the fact that drummer Larry Mullen Jr was due to undergo surgery, making him temporarily unable to perform with the band. And so the staging of *U2:UV* required the involvement of a stand-in drummer, Bram van den Berg, from the Dutch band, Krezip.

In August ’21, Joe O’Herlihy first flew to Germany to hear the HOLOPLOT system. “Bono said to me, ‘I want you to go over and check out this new sound system that [*Sphere developer*] Jim Dolan has been telling me about. And he says, ‘Do your best to blow it up…’

“It was the first time that they put 160 of these boxes together at the Congress Center [*in Leipzig*], an industrial exhibition hall, which they completely treated acoustically. I had my multitracks from the last tour that we did, the *Joshua Tree* tour [*2017-19*]. I brought an outdoor stadium show, I brought a dome show, so there were different environments.

“It was amazing. I was blown away by the thing, so much so that I stayed for a week! I must have driven the German engineers nuts. I was asking them, like, a million questions. Because this thing was so impressive. It was incredible. I had Bono’s vocal mic, and I plugged that in, and I could walk right up to the speaker, and shout and scream and roar into it as loud as Bono would. No feedback whatsoever. I mean, nuts!”

Previous Experience

U2 had already pioneered immersive performance in the round on their *Innocence + Experience* (2015) and *Experience + Innocence* (2018) tours. These productions involved a main stage and smaller, circular ‘B’ stage connected by a walkway the length of the arena floor, which featured a 96-foot-long double-sided video screen. During the shows, the band moved from one area to another, even performing from inside the screen.

At first, this presented an enormous challenge for O’Herlihy. His solution was to design an oval-shaped PA hung from the ceiling of the venue and directed down towards the audience. “Obviously, in a normal concert situation,” says Edge, “99 percent of the time, the bands are on stage with a PA left and right of the stage, and they’re blasting audio the length of the building. So what we did is we put speakers in the ceiling, so the audio was coming down to the fans.

“Not only did it avoid the issue of massive feedback as we were sort of »

The exterior of the Sphere is also a video wall, here projecting U2’s ‘space baby’ into the Vegas night.

» in the centre of the building, but also it solved the time-alignment issue that is everywhere. You know, particularly in festivals, where you're at the back of the stadium and you're looking at the video screen, you realise that all the lip sync is out, because the sound has taken such a long time to get to you. It's three milliseconds per metre of throw."

"That's probably what I would feel personally was my greatest achievement from a sound design point of view," O'Herlihy says. "The way the system was configured all the way around in that oval shape, I went left and right stereo image all the way across. So no matter where you were, you got the stereo image perspective of the way the show was mixed. And by doing it like that, you were no more than 75 feet away from the source sound."

"In this case," Edge adds, "because the speakers were very close to everybody, there was time alignment across the whole building. Not only did it solve some technical issues, it really improved the quality of the sound for the whole building. And we were really, really delighted with how that worked out."

"I mean, only with this band could you do it," O'Herlihy points out, "because there's a huge element of cost involved."

"Trying new ideas is definitely part of the spirit of the band and of our team," Edge states, while saying of O'Herlihy, "He's never scared to collaborate, he's always open minded, he's always interested in learning from other people. So, there's no ego. It's just literally he's got his eyes on the prize of the best possible concert sound."

Quantum Effects

Pre-production on *U2:UV* began eight months before the team could actually gain access to the Sphere. Instead, O'Herlihy and the band worked with a similar surround system at Victorine Studios in Nice. "It's a big film lot with huge sound stages," he says. "So the band were in Studio One, I was in Studio Four. In Studio Four, I set up a big grid system, and put speakers up in the grid… just hung them all over the place. I had 32 different destinations, which is the configuration we configured for the Sphere. So I was able to mix on my console."

At the Sphere, O'Herlihy uses two of DiGiCo's SD7 Quantum consoles. In France, he also managed to get his hands on a prototype of DiGiCo's SD8. "So, it was just fantastic," he says. "I mean, again, they're leading the field in the advancement, digitally. The new console is coming out, but it was a little too early for the Sphere shows.

"The simplest way to describe how we did the routing and configuration is that each channel has 40 auxiliaries. And with each one of those we put in a destination on the 32 speakers that we had. So you put one channel into destination 6 and 7, or 17 and 18, or 23 and 24, and when you brought the fader up, that would send the feed to that speaker through the auxiliary. That's the way we figured out how to do it, in our kind of mock-up version.

"The guys would come over from Studio One, and in Studio Four, we had Pro Tools running, so they could listen back to what they had just done, with the imagery and with the movement and all of that kind of surround stuff going on."

"That was useful," says Edge, "because we were really trying to understand the immersive aspect more than just the sound itself. We're going through a similar learning curve with Atmos. There needs to be a focus, there needs to be a discipline. So that's what was useful in France. We were exploring those parameters and finding where those lines were."

Pinpoint Accuracy

When O'Herlihy and the band finally got in the Sphere in the middle of 2023, the sound engineer was pleased, »

Joe O'Herlihy: 45 Years Of U2 Live

Joe O'Herlihy knows a lot about rock & roll gigs, and U2, having worked with the band since encountering them on a bill at the Arcadia Ballroom in Cork in 1978. He remembers being impressed by Bono's desire to connect with an audience, but not so much by the band's technical know-how. "Well, that was it," he chuckles. "They could barely hold their instruments, never mind figure out how to play them. I remember Bono jumping off the stage and encouraging the half dozen people that were there to turn around and to look at the band.

"Edge was one of these guitar players that struck me as being quite different. I remember him having a Golden Virginia [*tobacco tin*] on the floor, which was the switcher, and I was kind of going, What's this all about? He eventually had his little [*Electro-Harmonix*] Memory Man echo unit. He'd hit the switch that would engage it, and the 'Edge orchestra' began."

At the time, O'Herlihy was doing front-of-house sound for Irish guitarist Rory Gallagher. Despite his intention to quit the road and concentrate on running his PA company, U2's then-manager Paul McGuinness managed to convince him to go out on tour with the young band.

"I told my wife, Marian, 'Look, this Paul McGuinness chap wants me to do U2, what do you think of that?' And she said, 'Well if it's around Ireland, that's fine.' So I did that for a while. And of course, everything just escalated. From December 1980 to December '81, we spent seven months in America. We played in every telephone box you could find."

Through the subsequent decades, O'Herlihy has manned the live desk for all of U2's landmark shows, including the band's star-making performance at Live Aid in 1985. Meanwhile, his main memory of their key early days show at Red Rocks Amphitheatre in Colorado on June 5th, 1983 (recorded and filmed for the live album/concert movie *Under A Blood Red Sky*), mainly involves torrential rain.

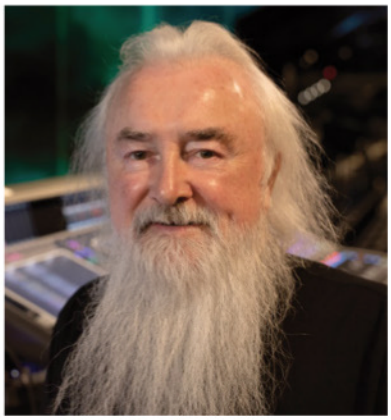


Photo: Ross Stewart

"Having a two-foot plastic bucket and emptying out gallons of water that had flowed down the hill into the front of house," he laughs. "Everything that could go wrong, went wrong, in the sense of the stage was flooded, front of house was flooded. At one stage Edge got a shock off the scaffolding because of his guitar and the grounding. But, I mean, in U2 tradition, and with the fighting spirit there, we got through it."

Working mainly in arenas and stadiums from the mid-'80s on, O'Herlihy pushed analogue live mixing technology to the limits, particularly in the '90s when U2's show began to incorporate the effected or programmed elements that were a feature of their sound from *Achtung Baby* on. "At several stages, I ended up with three Midas XL4 consoles," he recalls. "I had myself and two other guys operating them, to try and replicate what was going on in the studio."

Joe O'Herlihy and the band took the leap into digital live mixing in 2004, ahead of the 2005-6 *Vertigo* tour. "With DiGiCo coming on board and giving us the tools," he says, "from an audio reinforcement point of view, we could program into the console, and y'know, the magic appeared."

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» and a touch relieved, to discover that the preparation in France had served the team well. “Like with all of these things, you’re hoping for the best when you’ve done all of the homework,” he says. “Everything that we had planned for, everything that we had decided to do, had translated and configured correctly. It was amazing.”

Essential to the process was O’Herlihy’s tailored approach to using the Sphere Immersive Sound setup. “Where the stage is, where the band are positioned, directly above their heads is a thing called the proscenium PA system,” he explains. “How the approach was configured was that directly above Edge, we put his guitar setup. Bono was in the middle, so we set [*his vocal*] up directly above him. Bram was to his right, and Adam was to his left.

“The best way I can describe it is it’s like pictorial audio. So that when you’re looking at Edge, you’re hearing Edge, when you’re looking at Bono, you’re hearing Bono. If you’re down at the front of the stage, or if you’re on the floor, that directivity is absolutely essential, so that you’re hearing the guitar coming from over here.

“But it’s kind of everywhere,” O’Herlihy adds, “because directly above the proscenium system, we have another 13 speakers with immersive sound. So when Edge is using, let’s say, the shimmer effect, we can put it up there. The immersive thing is fantastic for effects and treatments. I mean, it’s great with Bono’s vocals and the backing vocals, and you can put reverbs and echoes and stuff like that there, so it feels it’s like a decorative thing.

“Then you go further back, to the very back of the building, and you have your surround system. There’s another 12 different speakers back there that give you the surround.”

Photo: Ross Stewart

The Sphere show has allowed U2 guitarist The Edge to explore stereo and spatial effects to a much greater extent than in any conventional performance.

Full Exposure

The live sound clarity now available to U2 presented certain other challenges for the group. When everyone in an arena can, in theory, hear every note played from the stage, another level of concentration is perhaps required from performers. “Oh, they have to be on their toes,” O’Herlihy laughs. “As Bono says to me, ‘I’m hanging my arse out there to dry. If I fuck up, everybody will know.’ I said, ‘Well, you’ve got to do your homework there, pal.’”

Faced with higher audio production values, U2 brought in their long-time record producer, Steve Lillywhite, who first worked with the band on their debut album *Boy* in 1980 and most recently on the ‘Atomic City’ single released in September, as Sound Consultant for the Sphere show. Lillywhite had only ever been involved in live sound once before in the past, and with less than satisfactory results. “Once, in 1979, Ultravox wanted me to mix their live sound,” he laughs, “and it was complete feedback the whole way through. I had no idea. It’s a whole different art form, live mixing to studio mixing.

“But what Edge thought with the Sphere was that it was this new sound system, and it might be good to have someone who was their studio guy to come and try and give a bit more of a studio feel to the sound. Also, because no-one had ever used this system before, it was a case of, ‘What can we do with it?’ So we tended to fill the sides with Edge’s shimmer, which is a huge part of the sound. And some of the keyboards off the [*Achtung Baby*] album.”

The Amp Pit

The stage for *U2:UV* is based on a Brian Eno-designed turntable and is remarkably bare, with no amps on stage. Instead, all of The Edge’s Marshalls and Vox AC30s, and bassist Adam Clayton’s Ampeg, Matchless and Fender amps, are positioned below the stage. “They’re cocooned in their own acoustic environment,” says O’Herlihy, “which basically means it can be as loud as it can be, or it can be as low as it can be, without being affected by anything. So, we get a really good quality sound from all of those. It is something that we’ve worked on and had to be very diligent about for quite a while to get that right.”

In addition, in terms of achieving specific guitar sounds, The Edge tends to switch between his traditional amplifiers and Universal Audio’s Ruby ’63 amp pedals. “He has kind of shifted a little bit to Ruby units,” O’Herlihy says. “But he’s always A/B’ing between them and the amplifiers. He’s experimenting all the time with stuff like that. But it’s pretty much the same kit.”

“We had a little bit of a problem with the guitar sounds sometimes being a little bit harsh,” says Lillywhite. “But, y’know, with some work, we got it. And really *Achtung Baby* has got the most guitar solos of any U2 album. It really shows Edge’s playing as much as anything.”

“I’ve got some stereo effects,” says the guitarist. “Normally, for me, I’m fairly mono, because in a conventional situation, there’s only a certain percentage of the audience that are in the sweet spot between the two speaker stacks. So [*with*] a lot of stereo stuff, you’re really catering to 10 percent of the audience. It’s not really that important.

"In this case, there's an opportunity for me to try some guitar sounds that really are distinctly stereo and immersive. Again, often simplicity is the best thing, and you don't want to get involved in gimmickry just because you can."

One major obstacle that had to be tackled during pre-production was Bram van den Berg's drum kit being the only loud, acoustic sound source on the stage. "If you were standing on the floor, especially," Lillywhite says, "if you heard the drums acoustically, you heard them before they came out of the speakers. Everything is processed through the system, and there's an inherent delay in the whole system. That's why one of the things that U2 hate the most, which is putting a screen around the drums, we had to do that."

Not everything needed to be reinvented for the Sphere shows. Bono, for example, stuck with his trusty Shure Axient, the digital handheld mic based on the SM58. "It's a bulletproof microphone for him," O'Herlihy says. "Because of all the antics he's been up to, year in year out, tour in tour out, y'know, hopping off everything he could find, trying to break it, we have a half dozen of those dotted around the stage in the event of him having a challenging evening, let's say."

First Night Nerves

Despite months of preparation, the team still didn't know exactly how the show would sound on the opening night. "One of the problems was that the band could never really listen to what it sounded like," says Lillywhite. "We would run a Pro Tools of them playing. But still, it didn't feel exactly the same as them playing. I mean, Bono could come out and we would run another vocal so he could hear what it sounded like out front, but you really needed the band playing to get what the actual sound was. So it wasn't a problem... but it was very nerve-wracking that first night."

"First night was a bit of a white-knuckle ride," Edge admits. "Because there's so much prep and yet, until the audience's in the building, it's just theoretical. And it was really a thrill, the opening night."

"Joe was fantastic," says Lillywhite. "He knocked it out of the park. He's very calm. He's very well suited to that job!"

"It was kind of a heart-stopping moment," says O'Herlihy, "to come to the end of the show, and look around and just see the incredible, incredible expressions on people's faces about what they had just seen and heard. For me, it was just sheer relief."

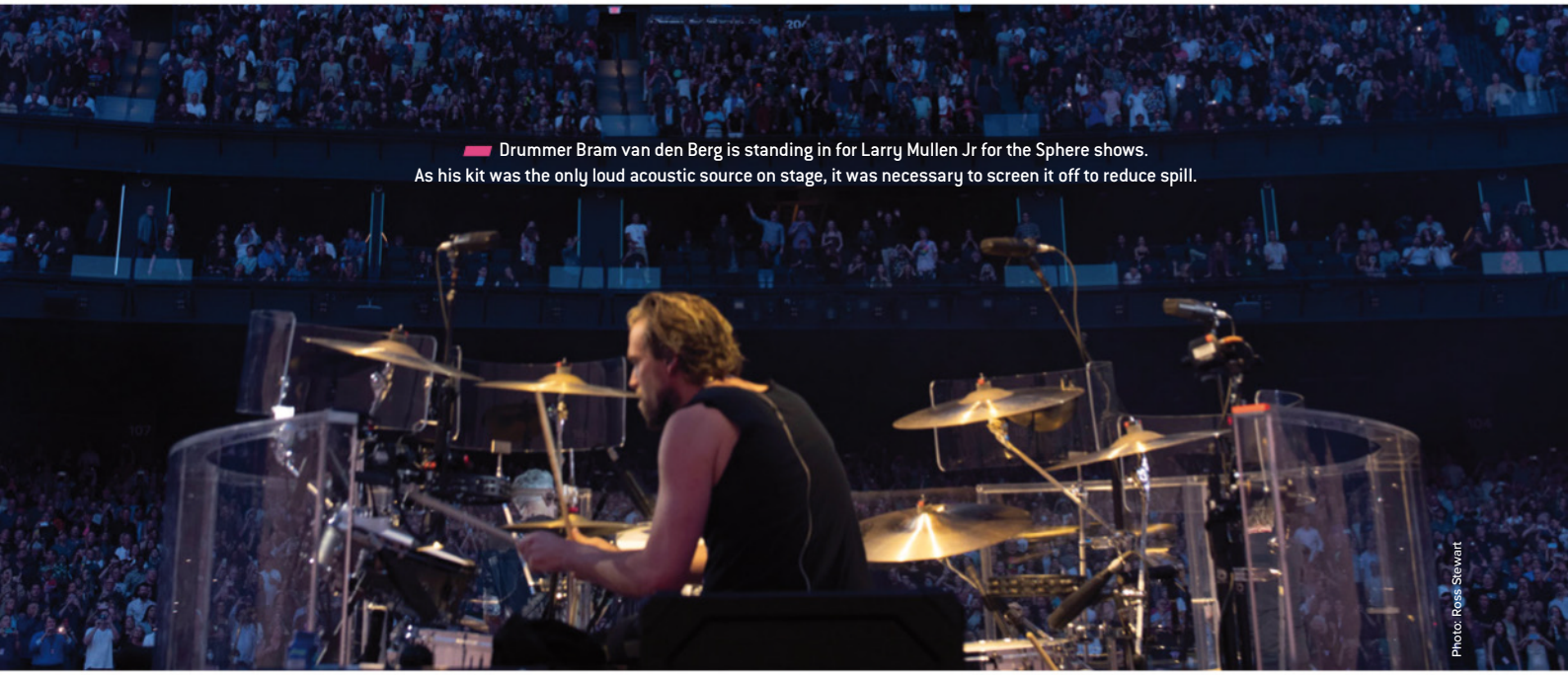
"Every night he learns something more about what we're doing on stage," says Edge. "So, you sort of hear the nuances of the balance. It's really going from strength to strength, I would say."

As to where U2 go from here, The Edge acknowledges that the Sphere show will be difficult to top. "Well, I mean, in terms of audiovisual," he says, "it'd be hard to imagine ever being able to do anything more immersive and elaborate and on this sort of scale. I mean, we could come up with some other kinds of show for the Sphere. We'll be thinking, I suppose, for the next number of years about touring shows and Sphere shows, because they are different things.

"But it has changed the game, I think, this venue. It really has created a new paradigm within live concerts. I'm fascinated to see what other artists might do with it. I mean, it's very much a blank canvas. So I'm sure there's gonna be some wonderful things created for this venue. And who knows? Maybe things created by U2 in the future."

"From an audio point of view," says O'Herlihy, "it is something that a lot of other acts should experience, and should experience the beauty of it all, and not being kind of fearful. There's nothing to be fearful about once you get your head around it. I mean, you have to look at it in a completely different light. But I'm sure there'll be a lot of people come after us. Because, I mean, it is the future."

With other venues mooted for Paris, Dubai and London, it looks like the ultra-tech Sphere show is here to stay. "Well, there is a version of the future that has U2 in nowhere else but the Sphere," Steve Lillywhite concludes. "U2 can keep playing *Achtung Baby* at the Sphere all around the world. But, knowing them, they'll think of something new."



Drummer Bram van den Berg is standing in for Larry Mullen Jr for the Sphere shows. As his kit was the only loud acoustic source on stage, it was necessary to screen it off to reduce spill.

Photo: Ross Stewart

MODULAR

Intellijel Sealegs

Eurorack Module

Intellijel already have possibly one of the most celebrated delay modules in Eurorack: the Rainmaker, a theme park of complex rhythmic comb-filter-flavoured goodness. You'll understand, then, why, upon hearing that a new delay had been announced by the British Columbian company, I was more than a little intrigued. The Sealegs takes the concept of delay down a divergent, arguably more trad route than the Rainmaker: it can capably assume the role of a workhorse multi-mode delay, but its flexibility and configuration makes it as creative and explorative as you could ever want a unit like this to be.

Weighing in at a comparatively compact 20HP (the 36HP Rainmaker was almost double the size), the Sealegs nonetheless offers copious CV control over a huge range of parameters and hence masses of modulation-based madness. At the heart of the Sealegs is a stereo delay with a choice of three different models: Tape, BBD (bucket-brigade) and Digital. There's also an onboard drive, a lush 1970s-inspired reverb nicknamed Fog (the sole maritime reference on the panel, I'm disappointed to report), a noise generator and a Freeze button. It's possible to place certain things, such as the noise and reverb, at different junctures of the effect chain and also to modulate pretty much anything, so spaces both realistic and fantastical are all in the offing here.

Along the centre of the panel is a row of sliders that might easily be mistaken for an EQ on first glance. These in fact pertain to an array of parameters, each with their own CV input and attenuator. Ostensibly, three sections constitute this row: controls for the delay proper, a modulation section and a 'character' section. In the delay section, the time and feedback parameters do exactly what they say on the tin — albeit in different ways depending on the selected delay mode. In Tape mode, for instance, changing the delay time will result in characteristic warping and pitch-shifting, while in Digital mode the stutter of a shifting buffer comes through. A switch underneath toggles the slider between ranges of long, medium or short delay times, with a maximum of eight seconds on the long setting and a minimum of just over three milliseconds on the short setting — well into flanger and chorus-type territories.

Between the mode and range switches is a third switch, for stereo mode. This offers Pong, 'T:W' (time and width) or L/R. Pong is more or less a conventional ping-pong delay, summing the left and right inputs to mono and sending them through the Sealegs' two delay lines in series, while T:W and L/R send audio through the two delay lines in parallel. In Pong and T:W modes, the Width slider adds a second, offset delay on top of the fundamental delay time, up to a maximum of 33 percent. That means, for example, that if your delay time is three seconds, Width's maximum setting will be four seconds. It's therefore easy to quickly create complex, double-delay rhythms, particularly when feeding the Sealegs a clock, in which instance it can sync. In L/R mode, the Time and Width sliders allow for totally independent control over the left and right delay lines' delay times, which is simply a pleasure to explore, and with the help of even a little modulation can create some truly dizzying soundscapes (or should that be seascapes?).

Next along is the modulation section, and in many ways this is the nub of the Sealegs' character. Beneath Rate and Depth sliders is a knob offering a multitude of behaviours including an excellent-sounding wow & flutter effect, positive and negative envelope followers and a low-frequency vacillator, similar to that on the Cascadia. On the right side is a smooth-sounding dual low-pass and high-pass filter, which once again is a welcome addition. The fact that Intellijel have chosen to give both the low-pass and high-pass filters their own physical slider instead of one multi-mode slider is a good choice, placing intuitive, hands-on tone control at my fingertips. Next to these is the 'Color' slider, , which is best described as a control for 'leaning into' the character of each delay mode. In Tape mode, this imparts drag, degradation and dropouts. In BBD mode it adds instability, noise and other idiosyncrasies. In Digital mode it introduces downsampling and bit reduction for an old-school sampler feel. All sound brilliant, without a weak link to be found.

A note on the noise: those familiar with the Cascadia will know its phenomenal choice of noise generation modes, which offer oodles of percussive, textural flexibility. That feature in many ways reminded me of how core a good noise generator can be on a synth, so I was eager to see what the Sealegs offered in this respect. Sure enough, the Sealegs' own noise knob has a trick or two up its sleeve, fading between options for pre-delay or post-delay hiss, or

a static-type crackle. While it does feel like a comparatively blunt instrument next to the Sealegs' other features, it can be modulated thanks to the assignable aux CV input to create swells of spacious hiss, percussive punctuation, or the subtle ebb and flow of vinyl-esque surface noise.

It's when all the Sealegs' controls come together that it really earns its name. With an abundance of undulating movement at hand, it can behave itself faultlessly or it can refuse to sit still; sliding, warbling and shifting in a constant state of oceanic flux and very much rewarding playing with sliders. Intellijel have balanced those poles of control and chaos impeccably here. The Sealegs is a reliably useful yet fabulously creative module, a more-than-worthy sibling for the Rainmaker. Outstanding. *William Stokes*

£ £399

W www.intellijel.com

Blukac Endless Processor

Eurorack Module

The Endless Processor from Kyiv-based Blukac does one thing. It takes a sound and sustains it to infinity. It's such a simple thing that initially, I felt a bit baffled and disappointed by it. Is that it? Surely there must be more than an eternally held note? But then it starts to undo you. Bafflement falls away and is replaced with mesmerisation. And when you resurface a few hours later after patching everything you own into it, you realise that this module is never leaving your rack.

As I say, the idea is simple enough. You plug in a sound source, Endless fills its buffer, and then you hit the big yellow button. At that moment, the sound is captured, resynthesized and smeared into the yawning forever. It's not like a sample that loops; this is the dismantling and rebuilding of sound waves that are clicklessly encircled into a constant stream of frozen sound. One way of looking at it is that it generates a drone where the tone comes from the thing you're playing.

A couple of unassuming but essential features open it up to vast possibilities. The first is that there are five layers of capture. So your drone invites some friends along, and soon a whole chorus of eternal voices are pulsing and swaying together. Each layer can be individually captured and replaced. A pair of Fade In and Fade Out knobs sets how quickly new captures emerge and old ones fade away. The other feature is that you have two channels of infinity to play with. So you can have two sound sources being captured independently, building up a huge soundscape of competing tones, again, utterly dependent on what you're feeding into either side.

To take you through the process, a simple scenario would be to put the output of a sequenced oscillator into the top. A small level knob mixes the input signal with the output, and a larger knob governs the overall output of the channel. The signal fills the buffer before the level control, so you don't have to hear the source at all, just the captured tones. The Memory knob in the middle sets the size of the buffer from very tiny to about 3 seconds. The effect of a long grab is quite interesting in that it might contain several notes or movements, creating a dense sound, but one that's completely frozen, with each harmonic or texture sounding together. To capture the sound, you select a layer and hit the Infinity button. A moment later, according to the time set by the Fade knobs, the sound emerges and continues forever.

I've used it to capture drone notes from a sequence, controlling the yellow button from a trigger with a slow fade so it comes in like a bowed cello. I've built up chords by feeding it specific notes over several layers and then modulating the output level to create rhythms and stabs. It can fade in an entire soundscape behind a moving sequence that's completely related to the sequence's notes. It can then fade away and be replaced with something similar and yet strangely different depending on how the source has changed. You can, of course, feed it unrelated sounds. I found that routing a microphone into the Endless enabled me to build some fabulous vocal textures.

The drones by themselves can be a bit dry, so having some effects to hand is essential. A deep reverb, a washing of chorus, or the shake of a band-pass filter will do wonders to animate the sound. There's no pitch-shifting in the Endless, which is perhaps one string missing from »

What's New

Good news for owners of the **Instruō** Arbhar granular processor; despite the groundbreaking module having been released around four years ago, the Glaswegian Eurorack developers have released a massive firmware update, dubbed "a complete rewrite of the entire codebase… to explore many more sonic and musical spaces." www.instruomodular.com

Qu-Bit have also been busy in the realm of granular processing, unveiling the predictably gorgeous-looking Mojave. "A granular sandstorm" is how the American developers describe it, with a portion of all proceeds going to the Joshua Tree National Park Association. www.qubitelectronix.com

Following **Erica Synths** taking over manufacturing of several iconic **Hexinverter** Électronique Eurorack modules back in January, the company have announced that the Hexinverter VCNO noise oscillator and Mindphaser complex dual oscillator modules are now available. www.ericasynths.lv

Not long after announcing they were discontinuing a spate of modules (including the lauded Basimilus Iteritas Alter drum synth) due to parts shortages, **Noise Engineering** have introduced Alia, a new module platform welcoming new iterations of Basimilus Iteritas and Manis Iteritas, as well as the brand-new Debel Iteritas Alia additive PM voice. "Debel can be gnarly if you want it to," say NE, "but it excels at bells, soft plucks, and it can be downright beautiful." www.noiseengineering.us

William Stokes

» the bow of infinite sustain, so patching into pitch-based effects can be very gratifying. As can adding modulation to the level controls to blend in some rhythms, tremolo or enveloped rises and falls. With a pair of LFOs you can crossfade between the two channels, which could simulate chord changes and texture variations.

The design is spacious, thoughtful and free from complications. You can build your whole system around feeding interesting things to the Endless Processor. Or you can leave it idly monitoring signals until the moment takes you to hit that button. Its ability to fill out space is extraordinary and very of the moment. What's for sure is that this remarkable module has an eternal place in my modular. *Robin Vincent*

£ £249

W www.blukac.com

Noise Engineering Xer Mixa

Eurorack Module

The Xer Mixa mixer is something of a step out for Noise Engineering, but it also takes the LA-based company back to their roots. "The story of Xer Mixer starts way back before you heard of Noise Engineering," reads the (very good, I should add) Noise Engineering blog. The 'problem child' that became Xer Mixa took almost a decade to develop, with around 20 scrapped versions preceding the product that I now have mounted in my case.

'A step out?' I hear you ask. Well, believe it or not, the Xer Mixa is not only the biggest Noise Engineering module yet, measuring a rack-hungry 32HP, it's also the very first to include a screen. This may come as a surprise, mainly because of Noise Engineering's stellar track record in the realm of digital — both hardware and software — and the strident design ethos underpinning the multifaceted (and essentially endless) Versio, Legio and Alia platforms. If you're not familiar with those then I'd highly recommend a visit to Noise Engineering's website to find out more.

With eight stereo inputs, two stereo returns and three sets of stereo output busses, Xer Mixa is a large-format, many-limbed beast. On first impression I couldn't but feel that considering the Xer Mixa's size, the mixer section somehow still feels a little cramped. In practice this isn't so much of a problem as I thought it might be — I never found myself hitting the wrong button or fumbling to pinpoint controls — but I can't help wondering if

a few valuable HP might have been shaved off the Xer Mixa's real estate with a few layout adjustments, as well as a little more workspace created.

The screen is where the Xer Mixa's deep-diving begins. I daresay that if you're averse to menu-heavy modules, you might be more than a little daunted by its workflow. Thankfully it's a very well-designed workflow at that, so don't be too put off. The Home screen presents high-resolution stereo metering (with both peak and RMS levels, usefully) for all three output busses, with the master volume adjusted by the Pan knob. This might have benefited from being labelled as such, since it feels a little counter-intuitive to adjust the master level with a knob simply labelled 'Pan', but nitpick I shan't.

Hitting one or more channel buttons calls up a display for that channel's (or selection of channels') volume, panning and A and B send level. Tap the Config button to then enter the Channel Configuration menu, which presents options for making the send levels pre- or post-fader — which I was jubilant to see — adjust the pan law and also adjust how panning information is sent to the A and B sends. All in all, that's some serious potential for precision editing, not normally seen on a Eurorack mixer in my experience. It's not only easy to set up a submix from either of the aux channels, it's also possible to mute channels from the master output, meaning it's possible to have three completely separate stereo mix busses going at once. Clever stuff.

Things get more creative with the Xer Mixa's generous amounts of remote control. MIDI control over volume, muting, pan and aux send level is available and, with the addition of the excellently-named Expando Expandi, CV control is too. The integration of Expando Expandi into the Xer Mixa's workflow is brilliantly done, with CV scaling and offset available as well. There is capacity for up to two of the expanders, too, so the scope for creative automation here is extensive.

As I said in my review of Cosmotronic's Cosmix Pro mixer, Eurorack mixers must draw the line somewhere in their functionality. Noise Engineering have all but torn that rulebook up, presenting a mixer that boasts functionality more commonly seen on those twice its size, and then some. It even has an in-built analogue levelling amplifier. It's almost certain to be the only mixer you'll foreseeably need in your setup — for Eurorack and perhaps beyond — which is handy, since, considering its price, it's also likely to be the only mixer you can afford for a while. Some aspects of its layout frustrated me from time to time, as well as its propensity for menu-diving, but all things considered this is one very powerful unit with tremendous staying power, not to mention an impeccably balanced creativity-to-functionality ratio. Noise Engineering can breathe a sigh of relief knowing their 'problem child' has finally made its way into the world. *William Stokes*

£ £995

W www.noiseengineering.us

CHRIS "WOODY" WOOD

BASTILLE | DRUMMER & PRODUCER
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DPA DDK4000

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Worth €4668

For this month's exclusive *SOS* competition, we've teamed up with esteemed Danish microphone experts DPA to offer you the chance to win one of the company's brand-new microphone kits. The DDK4000 must surely be one of the most extravagant drum-miking bundles ever: worth over £4000 and comprising no fewer than seven of DPA's premium microphone models, plus mounting and connection accessories, the DDK4000 has been carefully curated to be able to handle the most demanding of drum-miking applications with ease.

The 4055 is a cardioid-pattern electret mic made especially for bass drums. To that end, it features an asymmetric design for easy placement either inside or outside of the kick-drum shell, and it can handle SPLs up to a whopping 164dB. Unlike other kick-drum mics, it features a smooth, extended frequency response, allowing you to tailor the kick's sound with ease using EQ.

Also included is a DPA 2012 'pencil' mic, intended here for use on the snare drum but also suitable for any number of other instruments. Likewise, the two 2015 wide-cardioid mics in the DDK4000 are ideal as a pair of drum overheads, but will excel in any role where detail, clarity and pristine sound capture are called for.

To enter, please visit:
https://sosm.ag/dpa-comp-0124

The DDK4000 also includes three of DPA's world-renowned 4099 instrument mics. With their miniature electret capsules, these mics are designed to fit into even the tightest of spaces — an important consideration when miking a busy drum kit! They're also capable of outstanding results on a huge range of instruments, and the ones provided with this kit are the Extreme SPL variety, so will handle anything from tom-mic duties on a heavy metal kit to cymbal mics and jazz brushes with ease.

Finally this kit also includes all the connectors you'll need to plug the 4099s into your mic preamps, as well as three DC4099 drum mic clips for attaching the 4099s to your drum shells.

To be in with a chance of winning this amazing bundle, simply follow the URL shown, and answer the questions there, by Friday 2nd February 2024. Good luck!

Prizes kindly donated by DPA Microphones
W www.dpamicrophones.com/ddk4000

PSI Audio AVAA C214

Active Bass Trap

Standard low-frequency acoustic treatment needs a lot of space, but PSI's active traps fit anywhere.

PHIL WARD

If you're a regular reader of *Sound On Sound* you will be familiar with advice that goes something along the lines of: "Before you get too hung up on the sound of your monitors, you need to manage the influence that your room has on what you hear." The significance of that advice can't really be overstated, and even if you work in a room without a single item of acoustic treatment, having just a basic understanding of how the room might be imprinting itself on your work is better than not giving it a thought. If you want your work to translate successfully to other playback systems in other spaces, the influence of the room in which you mix is a pretty major factor.

When I mention 'acoustic treatment' you probably imagine I'm thinking of bass traps or absorption or diffusion panels, and yes, those kinds of products are part of the equation. But big items of soft furniture, bookshelves, curtains and carpets can also be effective tools in the battle to manage room acoustics. What all of these items have in common is that they are passive: they sit in the room and do nothing but absorb or diffuse excess acoustic energy, and as such, can't be targeted at specific troublesome frequencies.

As long as your room displays a reasonably short reverberation time at mid and high frequencies (say, less than 500ms), and isn't prone to major flutter echoes, those troublesome frequencies tend to reside below around 150Hz, where audio energy with a wavelength that equals some multiple of the room dimensions bounces between the primary room boundaries to generate resonant room modes. We've all experienced that phenomenon whereby the same mix sounds bass-light at one position in the room and bass-heavy somewhere else. This is because the two listening positions correspond to locations where room modes respectively suppress and boost frequencies that happen to coincide with elements of the mix. It's a cruel trick of architecture that room sizes and their resonant modes often correspond with the bass frequencies of music. If we all lived and worked in rooms three times the size, I perhaps wouldn't be writing this review.

Cancellation Culture

Low-frequency room modes are often dealt with using passive bass traps, which dissipate low-frequency energy either through plain absorption or a damped low-frequency panel resonance. Such passive bass traps can be highly effective, but they need to be pretty large, and so typically consume significant studio real estate. However, in 2016, Swiss monitor company PSI launched an alternative approach to bass trapping in the shape of the AVAA C20 active bass trap, which *SOS* Technical Editor Hugh Robjohns reviewed in that year's

PSI Audio AVAA C214
£2995

PROS
- Highly effective.
- Neatly designed.
- Easy to install and use.

CONS
- Expensive.

SUMMARY
PSI's AVAA C214 active bass trap is well designed and effective in use, and could potentially solve the room-mode problems that bedevil countless studios. It is, however, frustratingly expensive.

August issue (www.soundonsound.com/reviews/psi-audio-avaa-c20). The AVAA C20 looks a little like an active subwoofer, but it conceptually operates in reverse, reducing bass energy in the room rather than increasing it. It achieves this by employing a microphone to sense instantaneous acoustic pressure in the room, the analogue of which is then used to control the motion of an array of amplifier-driven subwoofer-style drivers in order to equalise the acoustic pressure. When the pressure rises with the arrival of a low-frequency peak, the drivers move backwards to reduce it, and if the pressure drops on the arrival of a low-frequency trough, the drivers move forwards to increase it. The principle is related to that employed in noise-cancelling headphones, just on a grander scale. Normalising the acoustic pressure means that low-frequency energy arriving at the bass trap, rather than being reflected back into the room, is effectively evaporated. The result is that the low-frequency resonant room modes are suppressed. Hugh's review of the AVAA C20 concluded that it worked very wel, and PSI have now introduced their second active bass trap: the AVAA C214. AVAA, by the way, stands for Active Velocity Acoustic Absorber.

The new AVAA C214 employs a somewhat different form factor from the C20, being cylindrical rather than trapezoid, and has a neat optional mounting bracket available that enables it to be hung from walls rather than just sit on the floor — although the latter is a perfectly acceptable mode of installation. The C214 incorporates very little in the way of onboard user facilities: just a mains power socket, a couple of configuration buttons and a status LED on the rear panel. It's clearly intended as a 'set and forget' device.

At 21.5cm in diameter by 63.5cm high, the C214 is very much smaller than any effective passive bass trap and is also smaller than its predecessor. The most significant difference between the old C20 and the new C214 is that the sensing and drive electronics within the newer product are entirely digital. As well as bringing significant technical benefits in terms of faster reaction time, that has provided PSI with the opportunity to develop a control and configuration app.

App & Away

The control app enables the gain level of the C214 to adjusted over WiFi, and for multiple C214s to be grouped. Changing the gain level increases or decreases the C214's sensitivity and consequently its effectiveness, but increasing the gain too far risks introducing instability (the C214 is a mic connected in close proximity to a speaker, so the potential for instability is inherent). Being too ambitious with C214 gain can trigger a feedback-like audible signature, though the phenomenon isn't quite as worrying as it would be in other audio contexts, firstly because it's a simple matter to wind the gain back a notch, and secondly because establishing the instability point is part of the setup procedure.

The AVAA control app lets you set the maximum gain for each AVAA unit, and assign them to different groups.

As with conventional passive bass traps, more is generally better, and multiple C214s are very likely to be more effective than a single unit. I was loaned four C214s to experiment with in my 5 x 4 x 2.2m studio room, and following the advice in the installation manual, I began by positioning one in each corner — corners being where the pressure changes due to low-frequency standing wave modes are likely to be greatest.

The next step was to connect them to mains power, switch them on and put them into WiFi connection mode by pressing and holding one of the rear-panel configuration buttons. Using the app it was then straightforward to get each unit connected to the studio WiFi network. I chose to group the four units into a front pair and a rear pair. The front pair were those located in the corners either side of my Neumann KH150 nearfield monitors, and the rear pair were those located in the corners at the other end of the room. Experimenting with the C214 gain setting convinced me that +3dB was a good maximum to guarantee stability; +4.5dB was borderline.

Measuring Up

Before discovering the subjective effect of the C214s in my room, I fired up FuzzMeasure to analyse their influence on the measured in-room frequency response. Diagram 1 shows the 1/12-octave smoothed, wide-band frequency response at the primary listening position with both KH150s playing. Note that the KH150s were previously optimised for the room using Neumann's MA 1 monitor alignment package (which, by default, adds a few dB of low-shelf gain below 100Hz). Measuring with the MA 1 alignment engaged meant that the C214s potentially had less to work with in terms of low-frequency modes, but it also raised the question of whether active bass trapping is still worthwhile once monitoring is optimised through room alignment.

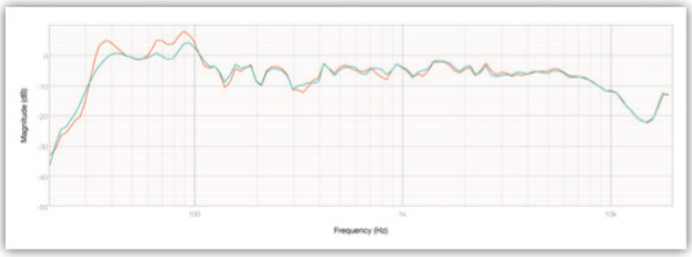

Diagram 1: Frequency response (12th-octave smoothed) of a pair of Neumann KH150s at the listening position, with AVAA engaged (green trace) and disabled (red).

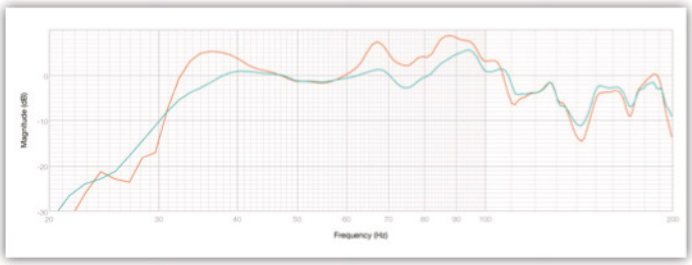

Diagram 2: Unsmoothed frequency response between 20-200 Hz, again with AVAA engaged and disabled (green and red, respectively).

And to answer both that question and that of the general effectiveness of the C214, the red curve of Diagram 1 shows the »

» frequency response with the C214s switched off, and it reveals evidence of modes at around 36Hz, 67Hz and 87Hz. The green curve shows the result of switching on all four C214s, and the difference is pretty clear: the low-frequency room modes are suppressed significantly and the low-frequency response is generally more linear.

Diagram 2 shows the same data but zoomed in on the 20-200 Hz band and now unsmoothed. Again, it's clear that engaging the C214s works well to suppress the room modes. But what, I hear you ask, if your budget doesn't stretch to four C214s? Diagrams 3 and 4 respectively show what happens when first the rear pair, and then the front pair, of C214s are switched off. Neither scenario is as good as the full-fat arrangement of four C214s, but two at the front is significantly more effective than two at the rear. To put some numbers on that, and looking at the 67Hz mode for example, four C214s reduced the mode by around 6dB, two C214s at the front of the room yielded a 3dB reduction and two C214s at the rear of the room resulted in a 2dB reduction. Remember, though, that these measurements reflect only the effect of the C214s at the primary listening position. The very nature of room modes means that a different measurement position may well result in different numbers. I also tried and measured a variety of other room arrangements of various numbers of C214s but found nothing as effective as putting one in each corner. For example, I had high hopes for two C214s at the front and two halfway down the room, figuring that bringing the second pair closer to the listening position might be more beneficial than locating them in the corners. It was pretty good, but still not as good as the rear corner location.

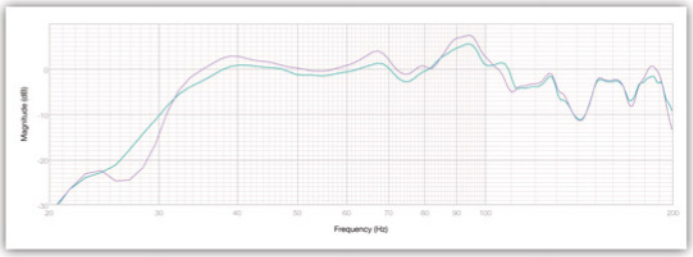

Diagram 3: 20-200 Hz response, comparing all four AVAA units (green) with only the front pair engaged (purple).

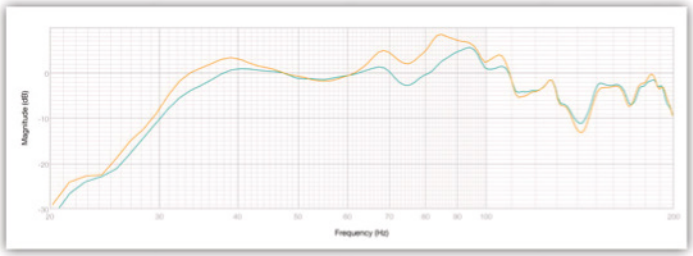

Diagram 4: Four AVAA units engaged (green) compared with only the rear pair engaged (orange).

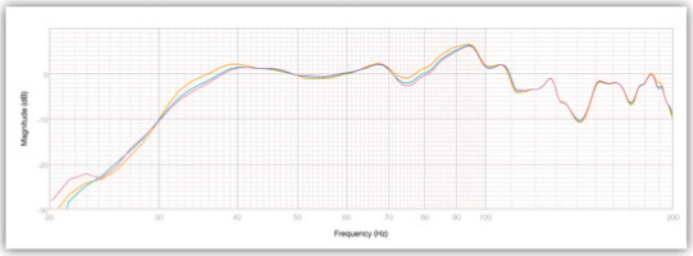

Diagram 5: Comparing all four AVAA units when set to 0dB (orange), +3dB (green) and +4.5dB (purple).

The last curve, Diagram 5, shows, just for completeness, the effect of varying the C214 gain. The orange curve shows 0dB gain, the green curve shows +3dB and the purple curve shows +4.5dB. As expected, suppression of room modes increases with C214 gain.

User Experience

So measurement reveals the objective effectiveness of the C214s in suppressing low-frequency room modes, but what about the subjective result? It's very impressive. Bearing in mind, firstly, that despite my studio showing low-frequency modes I've never found them to be particularly troublesome, and secondly that my monitors are already optimised for the space, the effect of the C214s was to very noticeably clean up and control the low-frequency character of the room. And cleaning up the low end has subjective benefits further up the music band: less low-frequency hash from room modes means voice-band midrange detail in particular is less likely to be masked. It probably goes without saying that, having established that four C214s working was the best option objectively, it also proved the best subjectively, but the improvement brought by just two C214s was very noticeable and worthwhile — even more so with the KH150 room alignment switched off.

Making low-frequency mix decisions is often one of the most difficult aspects of music production, and quite often that's down to the effect of room modes. The AVAA C214 is a technologically advanced, easy to use and brilliantly effective product that provides a neat solution to a genuine need, and in those terms it's a triumph for PSI. But it is expensive, and there's the rub, because even just two C214s command around the equivalent price of a pair of high-end nearfield monitors. Or to put it perhaps more starkly, a single C214 costs more than the pair of Neumann KH150 monitors I currently use in my studio. However, there are two ways of looking at this. One view might be that the C214 constitutes an unaffordable luxury, but an alternative view is that, bearing in mind just how much money is spaffed on studio gear generally, investing in something that has the potential to fix one of the most fundamental issues could be absolutely justified. Even so, sadly, for most of us AVAA C214s will remain an aspiration. In commercial studios, however, where work is plentiful, day rates are not insulting, and finances are viable, a brace of C214s may well absolutely be worth the investment.

ALTERNATIVES

PSI's original active bass trap, the AVAA C20, is still available and is significantly less expensive than the C214, so it may well provide an alternative option. A second alternative would be conventional passive bass traps, but while the cost will likely be an order of magnitude less than the C214, you'll need a lot of free space for installation and the result may not be as effective.

£ £2995 each including VAT.
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The Revolution Expands.

Years of performing boutique upgrade modifications to audio hardware taught us how to build our own high-quality interfaces from the ground up. Boutique component choices; stellar clocking; and the best headphone amps in the game are just the start of the Revolution offering. Now, the legacy of the Revolution 2x2 has grown... introducing the Revolution 6x6 and the Revolution EXP.

Revolution EXP

Expand your existing ADAT-compatible interface by adding 16 analog TRS connections and 16 channels of ADAT I/O. The possibilities are endless! Add hardware compressors, stereo reverbs, or other external hardware to your DAW workflow; record multi-channel drums; or connect a multi-speaker array to build a Dolby Atmos monitoring rig. Revolution EXP boasts the same Macro-MMC clocking tech from our standalone clocks and interface mods; use it as the primary clock for your entire rig; everything you connect to it will sound better.

Revolution 6x6

Beyond what you expect from an audio interface, the Revolution 6x6 is also a standalone DAC/ADC and primary word clock. Use the USB OTG connection to sample audio from your mobile device... or send audio from your laptop to your mobile device and stream your performances live online. All while treating your music with the highest audio quality imaginable. Sync your guitar modeler over S/PDIF for superior conversion and tone.

Join the Revolution at www.blacklionaudio.com.

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On Specs

Part 1: Why Specifications Matter

Specifications can be a useful guide when choosing equipment. Our new series in association with Audio Precision explains how to understand and compare the specs of competing products.

JOSHUA ISRAELSON

Credible audio-equipment and electronic-component manufacturers prepare product specification sheets with proper attention to accuracy. Manufacturers tend to be as forthcoming in their spec sheets as their competitive markets require. Yet when purchasing end-user equipment, customers — even relatively tech-savvy ones — can find various specs and different expressions for similar quantities confusing. This can make side-by-side comparisons of competing gear difficult and frustrating.

Judgement Days

When a company launches a new product development program, the internal spec sheet defines that product — its features, capabilities, and performance characteristics — to project contributors and stakeholders. Prior to product release, engineering and technical marketing staff will prepare a subset of the internal spec for release as a customer-accessible verified product spec sheet that identifies what purchasers can expect of the product.

In this fashion, one of the important roles a spec sheet fulfils is as a detailed expression of a product's value proposition: an account of what you're getting for your money. Knowing how to glean information from its tables, charts and graphs allows you to better assess a product's attributes, not just in absolute terms but also in the context of your specific interests.

For example, if you record spoken word projects such as podcasts, lectures, tutorials and audio for video, or acoustic music performances, you might value a mic preamp that features higher maximum gain and lower noise compared to a competing device. By contrast, if you're making death metal tracks, mic pre gain range and noise performance are less likely key concerns, because a lack of signal amplitude isn't going to be your problem. Other characteristics will likely occupy top spots on your must-have list.

Learning your way around various spec expressions equips you to correctly compare product performance in an absolute sense. Understanding the behaviours and traits behind the numbers enables you to assess performance claims in the context of your particular needs and interests.

Terms & Conditions Apply

There are some performance specifications that appear to quantify with a single value what is, in fact, complex multi-parametric behaviour. In these cases, the phrase "terms and conditions apply" is a handy one to keep in mind. Before comparing a spec on competing products, check that both spec sheets express the measure in the same terms, and that they specify the behaviour under the same operating conditions.

For example, power amplifiers' output power specifications don't have meaning unless they're accompanied by a stated distortion limit and load impedance. Amp makers virtually always express the output power rating in Watts, so having matched terms in this case is pretty much a given. But if one amp's output power spec assumes an 8Ω load and a competitor assumes a 2Ω load, the numbers aren't directly comparable, because the operating conditions don't match.

Similarly, meaningful output power specs must identify what the manufacturer has chosen as the maximum level of distortion. An amp specified with a maximum output of, say, 100W at 1% distortion may be a better choice than one rated at, say, 135W at 5% distortion if your interest is high-quality audio presentation. But the opposite might be true if you're selecting an amp to, say, drive the PA at a supermarket.

The Standard Drill

With the many specifications one can list for any piece of audio equipment and the huge number of possible

choices for test operating conditions, a situation could easily exist where every manufacturer chooses their own favourite combinations, making direct comparisons extremely difficult if not impossible. This is where test standards can establish agreed-upon spec definitions, test conditions and measurement procedures to yield consistent, repeatable and comparable test results.

Seemingly disparate organisations have developed a remarkable array of standards for audio applications. Some of these standards publishers are geo-centric, like the IEC (International Electrotechnical Commission) and DIN (Deutsches Institut für Normung). Others are application-centric, like the SMPTE (Society of Motion Picture and Television Engineers) and the EBU (European Broadcasting Union). Still others, like the AES (Audio Engineering Society), have broad interests in audio and draw from experts with wide-ranging backgrounds.

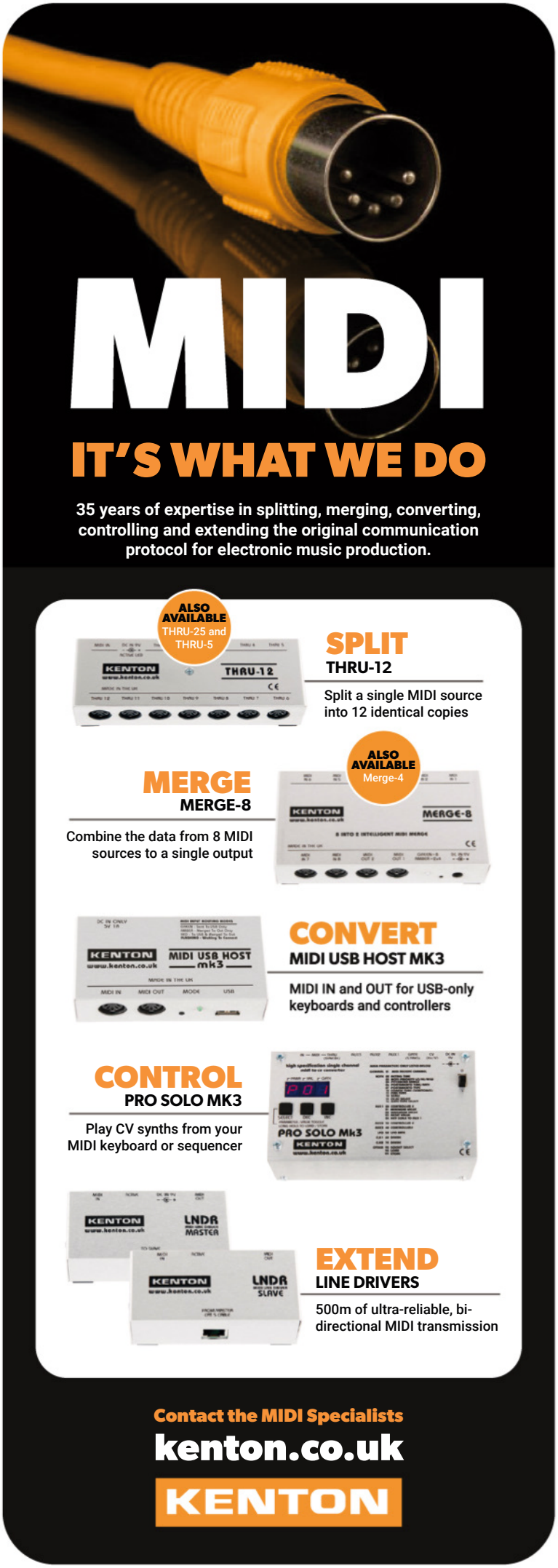
Less common, but worthy of note, are governmental agencies, which can regulate how equipment providers express certain specs (including reporting of test operating conditions) for listed gear selling into identified markets. An example is the FTC's (Federal Trade Commission's) Amplifier Rule Title 16 Part 432, which regulates output power claims in audio power amplifier advertising within the United States.

> "There are some performance specifications that appear to quantify with a single value what is, in fact, complex multi-parametric behaviour."

Since the end of the last century, the trend in technical standards across many industries has been to harmonise, or converge, specs covering the same topics for similar applications. This behaviour recognises that many equipment suppliers sell into a global market, so large savings accrue to those who can sell the same design across multiple territories or applications. This trend also benefits customers, because it increases the likelihood that individual specs will be comparable for competing equipment even if sourced from companies based in different countries. Greater choice usually implies greater competition.

When navigating equipment specification sheets, you don't need to know the details and requirements of quoted standards. You can consider standards as abstractions that ensure consistent spec expression, operating conditions and measurement methods for the parameters they cover. While you don't need to know the detail under the abstraction, you do need to confirm, when comparing competing devices, that both products' spec sheets refer to the same standard. Also note that standards evolve over time, so you want to be sure that the standard quoted in both cases is of the same revision or release date.

With thanks to Dan Knighten, Product-Line General Manager at Audio Precision Inc.



iZotope Ozone 11 Advanced

Ozone 11 builds on what was already a powerful and sophisticated platform with options including Stem Focus, the Clarity module and (shown here) Upward Compress in the Maximizer module.

Mastering Plug-in Suite

Ozone still offers traditional mastering tools — but its AI-fuelled options open the door to so much more!

JOHN WALDEN

iZotope's Ozone, which first appeared way back in 2001, played a huge role in the 'democratisation' of audio mastering — at a time when DAWs' in-built plug-ins were somewhat rudimentary, it was a powerful suite of software tools that anyone with a suitable computer and half decent monitoring could use. Of course, Ozone has evolved considerably since those early days but the latest version, Ozone 11, is the first major update since iZotope became part of the Native Instruments family. As with other recent versions, it's available in three forms to suit different needs and budgets: Advanced, Standard and Elements. All are available for macOS and Windows, and they can work standalone or as a suite of AAX, VST or AU plug-ins.

For a long time, Ozone required the user to have some understanding of the principles of mastering and how to apply them in practice. You can still apply such skills and understanding of course, but in recent years iZotope have been busy integrating 'AI assistance' tools that make it easier for less experienced users to benefit from Ozone's deeper functions. This is particularly obvious in the Elements version: in this one, the processing is essentially 'AI led', and the user is offered a compact set of macro controls to customise the software's recommendations. But my primary focus for this review will be the full-fat version: Ozone 11 Advanced. While this also makes greater use of AI than its predecessor, the user is trusted with more control and it boasts an intriguing array of new features and modules. Ozone is still designed primarily with mastering in mind, but it's definitely useful for more than that: in fact, there are lots of options that could easily be described as 'reverse mixing' — or perhaps 'magic', depending on your level of incredulity when Ozone does its thing!

iZotope Ozone 11 Advanced

£399

PROS

- Some impressive new modules.
- Stem separation expanded.
- Capable of far more than mastering.
- AI assistance very useful for the less experienced.

CONS

- Standard and Advanced have a learning curve.

SUMMARY

Ozone 11 Advanced is a deep and powerful processing suite suitable for mastering duties and more. The further integration of AI assistance will be welcomed by new and experienced users alike.

Make It Clear

A new module in Ozone 11 Advanced, Clarity expands on functionality in the existing Stabilizer and Spectral Shaper modules, and it's a similar concept to plug-ins such as Oeksound's Soothe or Baby Audio's Smooth Operator. It dynamically adjusts the tonal properties of your audio in real time to produce a more

The Clarity module is a very effective tonal balance tool for Ozone 11 Advanced users.

appealing tonal balance or, in iZotope's words, to "pull the blanket off" dull mixes but without sounding harsh. This is achieved by undesirable resonances being tamed and adapting the average response towards that of pink noise.

Clarity operates only on the signal above 300Hz, but you can narrow the focus using handles in the real-time frequency display. You can set the overall amount of tonal adjustment, as well as attack and release times for the processing, while a Tilt control lets you adjust the tonal balance of any adjustments for a darker (closer to a brown noise target) or brighter sound (nearer white noise). The different responses can be easily visualised on the display. With a mix that's already well balanced, Clarity might apply just a touch of icing on an already tasty cake, but it can also go a long way towards rescuing a mix or recording that's initially too bright or too dull, without the adjustments sounding forced.

Naturally, Ozone's AI assistance can configure Clarity for you, but you needn't rely on that. First, the module's preset collection includes useful options intended to address specific problems, for example Mud Removal and Tame Harshness. But also, whatever the specifics of the processing algorithms themselves, the controls all feel precise and intuitive, making this module very easy to operate manually too. And since Clarity is also one of the modules that's available independently of the Ozone 'mothership' (iZotope's term for the main Ozone plug-in), it's also pretty convenient — and addictive too, as a very controllable audio 'sweetener' — for use in a mixing context.

Ozone's Maximizer has always been a very capable tool for getting the final loudness of a signal exactly where you want it, and in the Standard and Advanced editions, a new Upward Compress control has been added to it. This allows you to boost quieter sections in the mix, and place more emphasis on those low-level details. Provided that you don't overcook things with this processor, it can be very effective. For example, applying only about 3dB of Upward Compress to a rock mix gave the track the greater sense of power, solidity and energy generally that I'd been looking for, and this seemed to be achieved without compromising the overall loudness or how transients and peaks came across. It's a welcome addition.

All Ozone versions now include the new Vocal Balance option in the Master Assistant. The Master Assistant has been suitably trained to identify appropriate vocal levels and if it spots a problem when it's 'learning' your audio, it will include the Master Rebalance module in the suggested signal chain. Set this module to target just the vocal stem, and then adjust the vocal level as required. You can, of course, make further tweaks yourself via the simple slider. However, if Ozone thinks the vocal level is already good, Master Rebalance is not applied and you just get a suitable tick of approval within the UI.

More Than Mastering?

As mentioned earlier, version 11 expands some of Ozone's capabilities to make it useful well beyond conventional mastering. The Master Rebalance module, mentioned in passing above and introduced in v10, already allowed you to separate a stereo signal into four 'stems' (vocals, drums, bass and 'everything else') and adjust the levels. But remarkably, for Advanced users, that stem-based processing has now been implemented »

Ozone 11 brings further AI-based assistance into the workflow with the new Vocal Balance slider within the Mastering Assistant display.

As shown here for the Low End Focus module, some of Ozone 11's modules now provide the option for separate processing of the transient and sustain elements of an audio signal.

» within the majority of Ozone's other modules, in the form of Stem Focus.

This can be engaged using buttons at the top left, allowing the whole signal chain to be switched between processing the full mix or, in real time, to dig within the spectral data and just apply the processing to the drum, vocal or bass. Want to hear additional compression only on the drums, EQ the bass, or apply a vintage tape effect to just the vocal? You can now 'reverse engineer' individual mix elements in precisely this way with Ozone. The only catch, really, is that you might need multiple instances of Ozone if you want to process both a stem and the full mix at the same time.

Standard and Advanced users gain a further 'dig into the mix' option. In addition to the stereo, left-right or Mid-Sides processing previously offered at module level, some modules now allow separate processing of the transient and sustain portions of a sound. And, of course, Advanced users can use this in combination with Stem Focus, meaning you could, say, apply EQ or exciter changes to just the transients of your snare, or different compression settings to the transient and sustain portions of your bass, or... well, you get the picture.

You can argue about the pros and cons of revisiting mix decisions in this fashion, and yes, of course it's still a good plan to get things right at source, but it really is remarkable to see this functionality in operation, and I have to say that I found it very effective. When it's simply not possible to revisit a mix project, these tools could well offer you a route to better results.

Among a host of other refinements included in this release, many modules now offer a 'delta' monitoring option (the ear icon located top left of the module window). This lets you hear just the changes — the difference between the original audio signal and the processed version — and it can lead you to some interesting insights, particularly when using EQ or dynamics modules, as it can make it much easier to spot if you're pushing the processing too far.

In an effort to improve workflow, a number of modules have been treated to a bit of a UI facelift. This includes the Master Assistant, Low End Focus, Spectral Shaper and Maximizer. By the way, graphical tweaks aside, I love that Low End Focus module: if you often struggle with creating a controlled, well-balanced bottom end in your mixes, you can quickly audition a range of presets that show you what various low-end treatments might do to improve matters.

Eleventh Heaven?

So, is Ozone 11 mastering heaven? I suspect it will have less appeal to those who are happy with their high-end analogue-based mastering chain. But for those who are happy working in the box, and those for whom DIY mastering is the pragmatic option, Ozone 11 definitely improves upon what was already a tempting proposition.

Given its combination of power and depth, Ozone 11 Advanced could seem intimidating to anyone new to the mastering process, but iZotope's integration of AI is starting to make a real difference in this regard — it can gently hold your hand while you explore mastering's pure science and dark arts. What's more, you have the option of starting with the compact, almost entirely 'assisted' Elements version, and then upgrading when you're ready to go deeper down the mastering rabbit hole.

But while Ozone 11 Advanced remains a great mastering tool, it is now so much more than that. Its individual modules can be put to very good use when mixing, and it allows you to revisit many mix decisions that would otherwise be baked in. This is powerful, deep, high-tech, sci-fi-style audio processing, and a lot of it bears a striking resemblance to magic!

Mastering Mastering!

iZotope liken Ozone's AI assistance to having a second pair of ears listening to your audio. I realise there's an assumption of trust built into that statement but, for those willing to take advice, I think the signal chains suggested by the Mastering Assistant process can be very educational. A good way to explore this is simply to inspect just how radically Ozone's AI is suggesting that you change your stereo mix. You will need to make good use of the Gain Match option (in the I/O panel beneath the meters) so that your ears are not misled simply by changes in loudness (bypassing the Maximizer module might also be useful), but then you can simply inspect each of the other modules in the chain to see how much processing is being recommended. The mini meters for each module (displayed in the signal chain strip at the top of the UI) suggest which modules are most active. You can then toggle Ozone's global bypass and/or toggle individual modules on/off and start to train your ears to hear exactly what changes are being suggested. Loudness aside, if those changes are fairly modest, then you can pat yourself on the back, because it means Ozone thinks you've created a mix that, in terms of things like tonal balance and dynamics, is in a decent place already. In this case, whatever Ozone's AI is suggesting is likely to be the subtlest of audio sweetening, which, in an ideal world, is what mastering ought to be about.

£ Ozone 11 Advanced £399. Standard £189. Elements £49. Included in Music Production Suite 6 (£579) and Mix & Master Bundle Advanced (£479), as well as subscription services. Prices include VAT.
W www.izotope.com

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UNLIKELY STORIES

The AKG DX11 Reverb Mic

The 1960s: a time of sexual revolution, radical politics and microphones with built-in spring reverbs.

SAM INGLIS

Today, live sound is more high-tech than studio recording. Whilst we cling to our vintage consoles and valve preamps, front-of-house engineers operate in a world of networked audio and digital effects. When big-name bands take their latest album on the road, the hire budget for the tour often dwarfs the cost of recording the album in the first place. But it wasn't always like that.

In the early '60s, even the best PA systems of the day were barely adequate. Sound reinforcement might consist of a single vocal mic, with amplifier and speakers of dubious provenance. Stage monitors were unheard of, and any control over the vocal sound beyond turning it up or down was a luxury. Then an Austrian mic manufacturer decided to bring some colour to this grey sonic world.

Spring Theory

Introduced in 1963, the AKG DX11 may have been the first active solid-state microphone, pre-dating the Fairchild F-22 / Syncron AU-7A by a couple of years. But it was not a capacitor mic, and it was not intended for studio use. Early transistor circuits couldn't compete with valves from the point of view of fidelity, noise performance or dynamic range. Their advantages were more to do with small size and low power consumption, and it was these qualities that AKG's engineers exploited to create the DX11.

By the early '60s, the use of effects in the studio was well established. Pop vocals on record dripped with slapback delay, plate, spring or chamber reverb — but there was no way to reproduce most of these effects live. Until, that is, someone at AKG had the bright idea of building them into the mic itself.

No advance in solid-state electronics would allow a plate, chamber or tape delay to be squeezed into a mic, and besides, AKG had no experience of building these devices. They had, however, spent several years investigating the acoustic properties of springs. That research would eventually lead to the unparalleled BX series of studio reverb boxes, but in its first public outing, it was paired with a moving-coil capsule and a battery-powered preamp to create the DX11.

Goes To 11

Either hideous or possessed of rugged industrial chic, depending on your point of view, the DX11 wasn't exactly a slimline mic. Even though the casing was made entirely of rigid, off-white plastic, the packed internals made it a pretty weighty affair. Not that this mattered much, because a mic with a spring reverb built in worked about as well handheld as you'd expect it to.

Some AKG dynamic mics from this period, like the D12 and D19, remain sought-after classics. However, the company also made a lot of cheaper models intended for consumer markets, and the DX11 supposedly repurposed the capsule from the D14, a plucky underdog that looks a little like the D12's younger brother.

Getting useful signal from a reverb spring requires a much hotter input than any microphone capsule can put out directly. Enter the DX11's newfangled solid-state electronics, which used germanium transistors to amplify the signal and drive the mic's single spring. These were powered by a single 9V battery. An on/off switch helped to preserve battery life when the mic was not in use, and the only other control was a wheel-like rotary which varied the level of the wet signal. This was thoughtfully positioned in such a way that the user could operate it with their thumb.

Like many AKG mics of the time, the DX11 was sold as an OEM product under several other brand names. It was even rebadged by Fender and sold into the US market as the FR570. But it never exactly set the world on fire, and was quietly discontinued in the early '70s. Was it too expensive to make? Was it ahead of its time? Or was it simply not very good? The answer, it seems, is a little from column A, a little from column B, but also quite a lot from column C.

The DX Factor

I've been curious about the DX11 since I first saw a photo of it, but not curious enough to pay the hundreds of pounds that some sellers ask. And when I finally found one, some work was required to get it up and running. As supplied, the DX11 had a captive cable to which the user attached their own output connector; depending on how it was wired up, this gave the mic either a 15kΩ or a 200Ω output impedance, but it was unbalanced either way. The cable on mine had frayed and needed replacement, so I deposited it with the long-suffering Stewart Tavener at Xaudia, who found space to fit a small balancing transformer, as well as a mini-jack socket that would allow the reverb to be used independently of the mic.

> "A mic with a spring reverb built in worked about as well handheld as you'd expect it to."

AKG's product literature for the DX11 claimed a frequency response of 50Hz to 18kHz, which seems a little optimistic. In practice, it has an obvious roll-off below 150Hz or so, which probably suits its intended role as a vocal mic, and a midrange presence peak that is fairly typical for a dynamic mic. The active circuitry is, as you'd expect, somewhat noisy and limited in headroom. The reverb, however, is surprisingly epic, especially when you consider that it's generated by a single, short spring. The decay time must be at least five or six seconds — perfect for those torch songs and ballads, less so for rock & roll. In mine, at least, it also has a characteristically 'ringy' tone, with a prominent midrange resonance that needs notching out with EQ. On the plus side, the thumbwheel that controls the wet level works perfectly, allowing you to fade the reverb in and out in real time.

Does it have a place in the modern studio? I've never heard of it being used on a well-known track, and doubt I'll ever record any source with the DX11 alone, but if you have space, there are certainly interesting results to be had by putting it up as a second mic on snare drum, percussion or vocals. It's more of a historical curiosity than a mic you'd use on every session, but it's different enough and distinctive enough that it might inspire something special once in a while. And, if nothing else, it's undoubtedly a talking point.

Cymasphere

Chord Generation Software

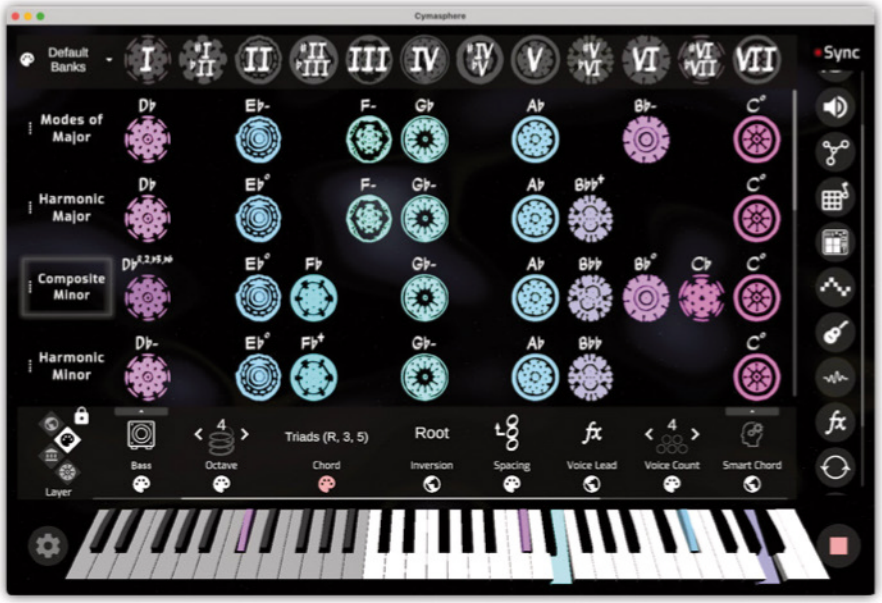
Cymasphere aims to make more complex chord construction available to all.

ROBIN BIGWOOD

Many musicians dream of a guru-level grasp of music theory, and especially that thorny business, harmony. If that's you, and several years of university-level study isn't on your roadmap anytime soon, this new app claims to be able to help.

Cymasphere is an app that runs in macOS, Windows, iOS/iPadOS and Android, and is developed by a small company based in Nevada, USA. To vigorously summarise what it does, it's a chord-focused MIDI note generator. Tapping or clicking one of the strange-looking coloured icons — a 'cymatic' — generates note data for a chord, which will be played either conventionally, strummed, or arpeggiated. In this sense, you can think of Cymasphere as a MIDI controller in its own right, designed to be played in real time, but it's also the toolkit with which to create different chord types in the first place. It's worth noting right away that the piano keyboard display is there only to display the notes being generated by the cymatics: it's not touch-sensitive or in any way interactive, as on most soft synths.

As to what you might actually play or trigger using Cymasphere, you have a few options. On all but the Android versions of the app there's a built-in SFZ-format sample player that comes bundled with a simple piano sound, with a further few ready to download. Many more possibilities open up if you use Cymasphere as a front end for virtual instruments running alongside it, via inter-application MIDI, or for your DAW, via network MIDI features (or in Apple ecosystems a dead-easy USB-cabled IDAM link). Then you might use it to play chords on soft synths or piano sample libraries. But intriguingly it could also drive an entire orchestral template (for example) in one go, because it can spit out individual chord notes on separate MIDI channels. Build up string, wind and brass sections from a good orchestral sample library, in a DAW, and after the necessary channel configuration Cymasphere will be able to play all the instruments (well, up to 16 of them anyway) at once, with intelligent voicings that keep individual lines moving smoothly between chords.



Intense Visuals

Visually the app is unusual, but it turns out to be perfectly logical and clear in practice. Running horizontally are multi-colour 'banks' of cymatics (ie. chords): some useful preconfigured banks are provided, or you can start from scratch and configure them as you like. The prevailing key can be changed at any time by tapping in the top-most row, the transposition bar, with the positions of cymatics beneath rotating to follow suit. It's also up to you whether you work with note letter names or the Roman numerals typical of formal harmonic analysis. Or indeed go totally free-form and turn off both, which is weird but can be bizarrely liberating.

Below the banks a 'dashboard' lets you spec chord construction: overall pitch, number of notes, inversion, spacing, and indeed the fundamental type or 'quality' (of which there are hundreds, user-configurable, from simple dyads and triads to fantastical creations straight out of the dreams of the most devoted Charlie Parker fan). Many of those settings can apply to a single cymatic, to a bank, to the currently visible 'palette' of banks, or even globally. Then to the right, a sidebar gives access to more general settings. Some have a bearing on real-time behaviour, like MIDI channel mapping, while others are more global in nature: there are buttons to access your online user area (with options to sync your Cymasphere creations, even cross-platform), a wide

Cymasphere
$179

PROS
- Draws on a massive harmonic vocabulary, not a bit dumbed-down.
- A novel, if currently somewhat niche, MIDI controller, with orchestration pretensions.
- Potential to open up new habit-busting harmonic pathways.
- User friendly, easy to use, with in-app tutorials and helpful online features.

CONS
- Harmonic complexity and open-endedness can be overwhelming.
- Current version limited to only a few basic chord playing styles.
- In multi-channel guise, with a DAW, a lot of configuration is required.

SUMMARY

An unusual and possibly unique chord construction toolkit, which also acts as a controller to play chords via MIDI. Quite usable already, with further development it could be amazingly useful for many different jobs.

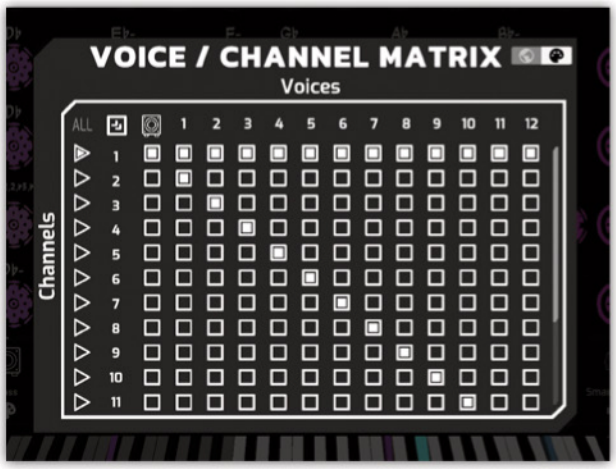
range of good built-in tutorials, and app colour themes. The 'sequencer' found there is more accurately an arpeggiator, with typical up/down patterns, note values and swing. There is a 'sequencer' pattern but it just lets you define a user arpeggiation by referring to each note within a chord as a number. Rests can be inserted, but there's no way to trigger more than one note at a time, or control velocity. This to me looks like an area ripe for extensive development, perhaps towards arranger or auto-accompaniment type features found in some workstation synths and sophisticated sample libraries. And indeed Cymasphere's founder, Ryan Johnson, confirmed that's the case: a DAW-focused scheme is in development involving a companion plug-in which will include piano-roll MIDI editing and more. It might even be available by the time you read this.

In Use

Cymasphere has few counterparts that I can think of. Some recent touchscreen Akai MPCs offer pages of ready-rolled chords that can be used to play their internal instruments. A few MIDI controller keyboards also have 'smart chord' features that let you dial in a key and chord type and then play chords with a single finger. But neither of these gets anywhere close to Cymasphere's harmonic sophistication and huge potential for user customisation, and certainly not its multi-channel capabilities. I also can't fault Cymasphere's execution. The appearance is unusual, yes, but not without its charms, and if the jazz 'real book'-inspired typography doesn't make many users feel at home, then the effective embedded tutorials should. I also found it about as robust as it could be when it comes to inter-application and network MIDI, and synchronisation, at least in the iPad+Mac setup I used for testing. Some scenarios, like using Cymasphere on an Android device to play instruments on another OS, require additional (though cheap) software and a bit of configuration, but it's all well documented on the website.

So there's a lot here that's impressive. Generating many notes from a single finger touch is a great trick, whether you're using it for fleshing out orchestral parts or playing piano chords. Juxtaposing lots of different, interesting chords, sometimes in remote keys, also has the potential to help break songwriting and composing habits. Happy harmonic accidents happen frequently.

The flip-side, though, is that, in this version at least, the number of playing styles is limited: only block chords, strums, or monophonic arpeggiation. You'll have to wait for the as-yet unreleased sequencing functionality to achieve true orchestrations, as opposed to automated orchestral chord voicings, and even most typical piano or guitar accompaniment figures. You also have to work at making Cymasphere-generated parts expressive. By default all notes are produced at a fixed velocity. A dynamics feature generates some random velocity deviation and tempo-sync'ed cyclical swells, but to get real-time control of much else involves connecting a hardware MIDI controller and remapping incoming data. That seems oddly redundant on an iPad, for example, when its multi-touch capabilities aren't otherwise utilised at all.



The ability to send out constituent chord notes on different channels is the basis of Cymasphere's ability to play an orchestral scoring template in one fell swoop, and it's all easily configurable.

One other thing that ate away at me a little during the review period was the feeling that it's not entirely clear who Cymasphere is really for. It's not that the app isn't user-friendly, or that new users couldn't get something out of it straight away. It's more that Cymasphere puts technical harmony terms and other music theory concepts so front and central that to get the most of it, and use it with real intention and control, you'd want to already have a really good understanding of them. Which is paradoxical, because if that were the case you might not actually need the app in the first place... In that light, a mode that more closely resembled those MPC-style chord collections I mentioned earlier, and which hid a lot of harmonic and configuration complexity, might be more useful to beginners and instinctive writers working in pop-leaning styles. Apparently something like that is on the cards, along with additional tutorials that walk through setting up specific chord progressions, as well as general music theory lessons. With those enhancements I imagine Cymasphere could be a much more enticing proposition for a broader range of users.

Conclusion

Ultimately, Cymasphere is a fascinating app, and at times perplexing too. If chord formation is your thing, it could prove to be a brilliant harmonic playground. Orchestral sample library users might also rejoice in the multi-channel voicing features. For others, the appeal and utility could be less immediate and obvious. It's full of promise, though, and a free demo version and flexible ownership options make it easy to try out: well worth a stab.

£ $178.80 or subscription $70.80 per year/$7.20 per month. Prices include VAT.
W www.cymasphere.com

Cymantics

If you're wondering about the name Cymasphere, it comes from 'cymatics', the term coined for modal vibration phenomena, as when sand scattered on a metal plate (for instance) forms into distinct shapes as the plate is bowed or vibrated by other means. The app's chord icons resemble some of those shapes, reflecting the different qualities of notes and chords. For more on this Google 'Ernst Chladni' and be ready to go down an interesting historical/scientific rabbit hole that somehow links music with quantum mechanics.

While we're on the subject of names, it's also worth noting that in Cymasphere only one type of chord notation is currently available, with minor chords shown by a dash and diminished with a superscript circle. Adapting to some of the other systems in common use (variously 'm', 'min', 'dim', 'aug', alongside some other symbols) is apparently a forthcoming feature, which will be great to see.

TalkBack

Afrodeutsche

WILLIAM STOKES

Henrietta Smith-Rolla, aka Afrodeutsche, bridges musical worlds like few others. Some may recognise her voice as host of the *People's Party* show on BBC Radio 6 Music, a slot she took over from Iggy Pop, and others will remember her work on the *Forced To Flee* series commissioned by the UN, alongside Max Richter. The crossing of borders has always been a subject close to home for the British-Ghanaian-Russian-German composer, DJ and producer, whose moniker was chosen after an exploration of her heritage.

"I started looking for my father, who I'd never met before," Henrietta explains. "He had German heritage, so after leaving Ghana he ended up with a scholarship in Germany as an artist. While I was doing the research, looking for an African-German, this word kept reoccurring: 'Afrodeutsche'. My German friends agreed it was a historical word, and one that they don't use. And I did not want to appropriate the word. But what started to happen, when I was playing in Germany, was that African-Germans would come and they would say, 'We're calling ourselves African Germans. We're saying we are Afrodeutsche!' which was bonkers! Young people, like, third, fourth, fifth generation."

At the moment I can't stop listening to

Radiohead. *In Rainbows*. I think people are starting to understand that I am an uber Radiohead fan. I tend not to listen to Radiohead's albums when they release them. I come to them a few years later. And I love that. That was the same with *In Rainbows*. I can start it and I will listen to every single track in the order. I appreciate the 'order of service'. I appreciate how it begins. I appreciate how it lulls in the middle. And I appreciate that it becomes so much more indie by the end, you know? The combination of electronics and indie... I play air guitar to Radiohead! I was in South Korea a few weeks ago. And of course, we did karaoke. And I chose 'Weird Fishes'.

It's interesting, I think people assume that when you make music for a job, you must listen to lots of music, all of the time. I have to put on so many different musical hats, so often I just focus on one piece of really good music, and it will be on repeat. It will carry me through all of those things. My world is music and sound. I'm a sponge. If I'm listening to things which are great or are by people I admire, I will creatively pull that into my work. And that won't be by choice. That will be because I'm absorbing things. So, I try to listen to the opposite of what I'm doing.

The project I'm most proud of

I think it has to be the last commission I had. It was called *Afrodeutsche Presents Psalms*. And they're songs of love. The preparation for this show was the hardest I've ever found it, but I managed to get together a team of the best of the best. I had to teach myself to play my songs again on the piano — and teach myself how to sing and play piano! Then there was the composition for full strings, and then working with Michael England on the visuals. Working with the MIF team as well, the responsibility was insane! They'd opened up this brand-new building. And there I am, doing, you know, costume changes! The whole thing. I'm proud of it because I put everything on the line for it.

The first thing I look for in a studio

That's a very interesting question, because it's a very rare occurrence that I do walk into

Photo: Kasia Zachanko

anyone else's studio. I mean, for me, it's the chair. Because I'm like, 'Where am I in this space? This is your space.' I'm extremely spatially aware of what is around me. Often my brain processes light in a very interesting way — I have something called Irlen Syndrome, where my brain essentially registers the patterns in light. So, often, I'll have like auroras over my eye, because my brain has been stressed by how much light it's having to process. A good example is a fluorescent light, which is actually going on-off-on-off-on-off, and my brain will register the on and off. So you can imagine, my brain just gets 'hot'. Full. And I have a meltdown. So I have trained myself to find the right, most comfortable spot in a studio or in a workplace, so then I can function. If I don't, my brain switches off.

Other people's studios really confuse me. I mean, obviously everyone works in their own way. Often I find other people's studios will have huge desks. And a lot of those channels on those desks are not being used! There was a season where I came to understand that about bigger studios, so I wasn't intimidated by them. I think that was the big change: making sure that the studio isn't a place that actually stops you from writing or working just because you're intimidated.

The person I would consider my mentor
There are a few! I've collected them over the years, and I am pleased to say that they are dear friends. The first person is Michael England: he is an artist based in Manchester. First of all, we have similar brains, similar interests. He would just give me music, regularly. He knew what I would be into. He was the first person to play me Drexciya, over 20 years ago, I remember sitting in that lounge, just amazed. I was stunned, thinking, what? And he knew it! He knew that that was my dream music. And there it was. Endless music. We share a passion for sound and science, and art, translating sound. Putting those two things together. So I've worked on scoring a few of his projects. He works at SODA [*School Of Digital Arts*] at Manchester Metropolitan University, and his work is my favourite. So it was just incredible to invite him to put the visuals together for the show. Gareth Carberry is, like, the hip-hop side of me. We will dissect everything, and we just have that much fun. Whenever I see him, I learn something new. Gareth and Michael, you will not find anything about them online. They've managed to do that. They stay away!

My go-to reference track or album
I have to listen to so many different genres of music, so that's a really difficult question for me to answer. Other than Radiohead, I don't really have any other place that I go, to almost 'wash', sonically. I think, in terms of classical artists, Arvo Pärt would be one I will jump into when I need classical that isn't necessarily Chopin. Even though Chopin and Mozart I love. Yeah, I would say it's more artists than albums; in the sense that I will jump into parts of their catalogues, if that makes sense. Radiohead, Chopin, Arvo Pärt. And this last one might trigger some people, but [*whispers*] also Hans Zimmer!

My top tip for a successful session
For me, working in Ableton, setting up the session well means I have a successful session. If I'm working on composition and scoring for something, having every instrument ready to go, just having everything balanced, as much as you possibly can, against each other. Often, when I'm doing orchestral stuff, if the piece is going to be played in a room, I will place each instrument where it sits in that orchestra in the room. There will be a reason why certain instruments aren't next to other instruments, and why certain things are at the back and other things are at the front.

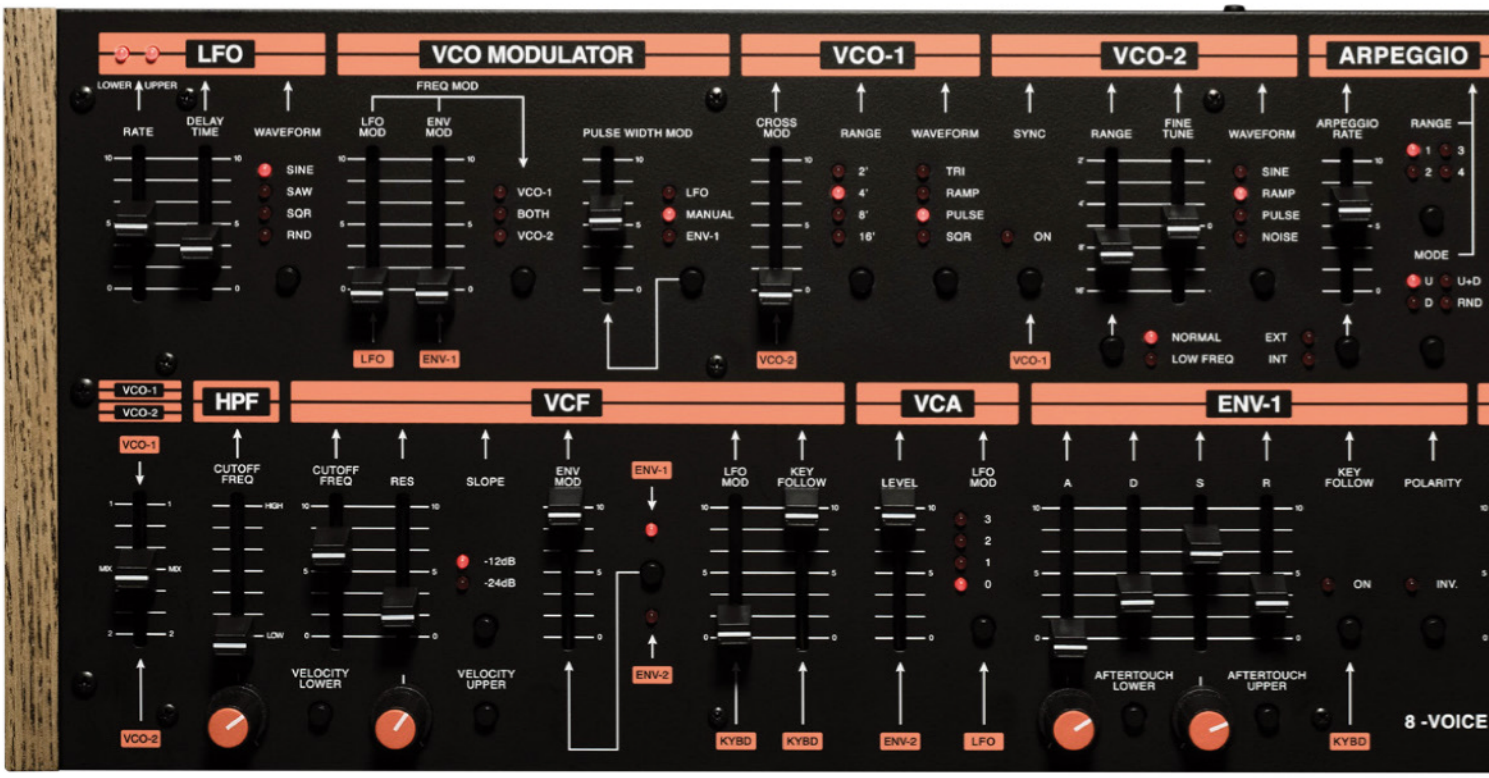
The studio session I wish I'd witnessed
There are quite a few, but I'm gonna say the Beastie Boys. Something I've always said about the Beastie Boys is that I'm just so glad that they met each other and they became friends! Like, it's one of those groups. And I'm so glad. And you can hear it in the music. But when they introduced Q-Tip something transformed. The session for 'Get It Together'. Like, come on! I think the Beastie Boys are just full of joy and humour. So I just imagine sessions with them are hilarious. Constant jokes and constant conversation. And I'd love to have seen Q-Tip in the room with them. I imagine that he is as funny, if not funnier.

The producer I'd most like to work with
I think, JLIN. She's an artist that I saw many, many years ago in Manchester, at a place called Islington Mill. And she was doing a live set. She makes, I guess, music in the footwork, trap kind of world. And that's the only thing I knew about her. And I was so transfixed by her setup. She had controllers all over the place. And she was using CDJs. It was like, 'How is this happening?' You know, there wasn't, there wasn't anything analogue going on, but it was just controllers with... things! She'd obviously mapped everything. I was like, 'I think I love you.' And from that day on, I would just listen to whatever she released. And I've heard her production become so completely hers. So executive. She's moved away from the genre. And made her own genre out of footwork. It reminds me of Drexciya and Dopplereffekt, in that it's contemporary classical music. It's just electronic sounds rather than, you know, the cello.

The part of music creation I enjoy the most
I'm going to give you two. I think, the most satisfying part of when I'm working for someone else, when I'm working on a commissioned composition, is when I am rendering the stems. It's like, the creation has been done, the time has been spent, and now it is at the point where I'm handing this over to a music editor to lay across this picture, this film or this documentary. And it's like, you realise how much writing you have done. How many cues you've written. Hundreds of cues, all with timecodes that span eight digits long!

The second one is just the way it's one of the only places I have for complete freedom. When I'm writing my own stuff. It's like, freedom. When I first started making music was I would have that voice in my head that would say, mid-session, 'That sounds too much like Carl Craig! They'll know that you listened to that! They'll think that you copied it!' So one thing that I taught myself was to respond to that voice, 'No.' I started saying, out loud, when that voice cropped up: 'No!' It worked. Just saying no. And that's where the freedom is.

The advice I'd give myself of 10 years ago
I think, 10 years ago, I couldn't have expected to be where I am. I think I would have said: persevere. That would be the advice I'd give my younger self. Because I wouldn't be here unless I had persevered. But it would have been nice if I'd known that then!



Black Corporation ISE-NIN

Analogue Polysynth

Have Black Corporation managed to capture the spirit of the Jupiter-8 in a 19-inch rack?

GORDON REID

Black Corporation have made a small but solid name for themselves in recent years, releasing desktop and rackmount analogue synthesizers (primarily polysynths) that are inspired by some of the most sought-after classics of the 20th Century. Recently, their focus turned to the Roland Jupiter-8, a synth whose reputation and value seems to increase as each year passes and — in a world replete with digital emulations — have released an homage with a true analogue signal path.

The Body Of A God?

Originally supplied in kit form and now available as a built instrument, the ISE-NIN is a light but solid desktop module that also comes with rack ears so that you can mount it in a conventional 19-inch rack. Its panel looks much like the programming section of a Jupiter-8, albeit one sawn in half with the two chunks mounted one above the other. Nevertheless, to save size and possibly cost, all of the switches — whether the original knobs, toggles or the row of coloured buttons that did so much to define the look of the Jupiter-8 — have been replaced by small buttons. Obviously, there's no controller panel to the left of the non-existent keyboard, although all of its facilities (and much more) can be obtained via the master section and MIDI.

Like the Jupiter-8, it's an eight-voice polysynth, and each voice comprises two oscillators, a high-pass filter, a dual-mode low-pass filter and an audio amplifier. Shaping is again performed by dual ADSR contours, although these are now digitally generated. As before, the voices can be accessed in three Key Modes. Whole mode assigns all eight to a single layer called Upper. Split mode assigns four voices to the Upper layer and the other four to the Lower layer and allows you to play these as independent synths either side of a user-defined split point. Dual mode again assigns four notes to each, but layers them for bi-timbral sounds with four-note polyphony. You can transpose either layer by ±36 semitones to make it 'fit' the areas on the keyboard that you define. Each layer also includes a single, digitally-generated, sync'able LFO that affects all of its voices equally.

In the upper row of controls, the two VCOs per voice offer the same range of waveforms and footages as the Jupiter-8, with the same PWM, cross-modulation, sync and VCO2 low-frequency options. Likewise, the LFO and VCO Modulator are reproduced correctly. The first significant differences appear to lie in the arpeggiator controls but, despite the visual changes and the apparent loss of the Hold and Latch functions (which have migrated to combinations of buttons), its function is largely unchanged. Hmm... that isn't quite right. Arpeggios on the ISE-NIN are velocity sensitive, you can choose two patterns simultaneously (one for Upper and one for Lower), they can have separate rates, and you can synchronise them to MIDI Clock.

Dropping down to the lower row, you'll find the VCO mixer and the high-pass filter. This is, perhaps, where you'll find the first additional physical control on the ISE-NIN but, although it appears to be associated with the HPF, it's not, so we'll return to this shortly. Next comes the dual-mode (12dB/oct and 24dB/oct) low-pass filter section, which again echoes the original's. I suspect that it would have been tempting to push the feedback gain in the ISE-NIN's filters to make them capable of oscillation, but the company resisted this and they respond as a Jupiter-8 aficionado would expect. To the right of the filter section lies the VCA and the two contour generators, which, again, recreate the Jupe's facilities. The final knob is a master volume control.

So let's now return to that knob sitting beneath the HPF slider. It turns out that this is the first of a set of four such knobs and four associated buttons that allow you to determine the velocity and aftertouch response for each of the Upper and Lower layers. Routing is simple; press one of the buttons and you'll be presented with a page on the tiny OLED display that allows you to select up to five destinations and their associated amounts. I counted 27 possible destinations for velocity (all of the faders plus Master Tune) and 25 for aftertouch (the same set minus the two release faders). This means that you have two sources, 10 slots and 27 destinations per layer. It's not as all-encompassing as the performance options within a digital workstation's modulation matrix, but there's a huge amount you can achieve here. Before moving on, I should note one anomaly. The manual says that the Upper/Lower Balance parameter is a destination, but it's not on the review unit. Perhaps this becomes available with a firmware upgrade (see box).

The menu architecture comprises 13 sections. Some of these include just one or two functions, while others offer numerous parameters, but none are deeper than a single layer, which means that, should you be tempted to dive into them, you'll bang your head on the bottom and risk crushing a few vertebrae. Among the options on offer lie a set of calibration routines that allow you to tune the oscillators, the cross-modulation and the filters, as well as the responses of various knobs. Elsewhere, you can customise the voicing by choosing the ranges and key sync of the LFOs, by determining the minimum and maximum times for the contours' attacks, decays and releases, and by setting the maximum portamento/glissando time. None of these options detract from the character of the Jupiter-8 emulation, but they all extend the voicing in useful ways.

Black Corporation
ISE-NIN
£4699

PROS
- It really can sound like a Jupiter-8 — in other words, gorgeous.
- Anyone familiar with the original should feel at home right away.
- Poly-aftertouch and MPE take it to places where no analogue Jupiter has ever gone before.

CONS
- It doesn't have a premium feel, the screen is very small and the menus are not always as clear as they should be.
- Upgrading the firmware is far from straightforward.
- It uses an external power supply with no stress relief for the cable.
- It's expensive.

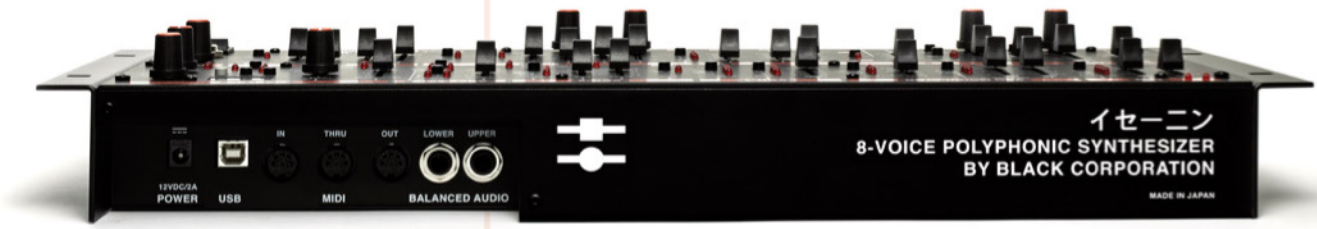
SUMMARY
Although it's encapsulated in a box that's a tiny fraction of the size and weight of its inspiration, the ISE-NIN has the soul of a Jupiter-8 and adds facilities and flexibility that may have seemed fanciful in 1982. It's expensive, but I suspect that some people will be able to justify the outlay. For others, the sound of a Jupiter-8 generated by an analogue synth will remain almost as far out of reach as ever.

Other features that I haven't yet mentioned include micro-tuning, optional configuration of the outputs so that the Upper and Lower layers are mixed internally or sent to separate outputs (the latter of which turns the ISE-NIN into two four-voice synths when played in Split mode), optional note stealing, plus low-, high-, and last-note priorities with the choice of single- or multi-triggering. This is all good stuff. Then there's the analogue drift control. You might ask »

The Rear Panel

The simple rear panel boasts balanced quarter-inch outputs for the Upper and Lower layers, which obviates the need for the Jupiter-8's XLR sockets. (A headphone output that mirrors these sits on the top panel.) MIDI is provided by a full set of 5-pin in, out and thru sockets, as well as USB. There are no analogue control inputs.

Unfortunately, the 2.5A power supply is external and, as is becoming increasingly common, it's not one that you'll find at a local store should you lose the original. Furthermore, there's nothing to stop the barrel plug from popping out if you accidentally trip on the cable. Come on guys... how much would a little plastic restraint add to the build cost?



» why this is included; after all, this is an analogue synth! The answer is to add a bit of additional drift into the tightly tuned oscillators, and some variation into the contours. Used sparingly, it can sound rather nice. Another touch that deserves mention is the use of the Shift button while adjusting a slider to update both layers in a Split or Dual simultaneously. This is great if you want (for example) to slow the attack or extend the release of a bi-timbral sound in Dual mode. Finally, let's return to those lovely, coloured buttons that, amongst other things, select the Key Modes as well as the Assign Mode (Poly1, Poly2, Solo and Unison), and the panel mode (Upper or Lower) on a Jupiter-8. Clearly, these have gone walkabout, but their functions have all migrated to the Master panel. It doesn't look as pretty, but everything is present and correct.

"Programming my favourite Jupiter-8 patches on the ISE-NIN, adding lots of performance control, and then playing the results using a Roli Seaboard Rise 2 gave new life to sounds that I've loved since the 1980s"

The Voice Of A God?

Braving the monsters (well... OK, actually the mice) that seem to take their winter holidays and dance the nights away in my nice dry storeroom at this time of year, I liberated my Jupiter-8 and set it up next to the ISE-NIN and its controller. Once both synths had settled and I had put them through their tuning procedures, I performed all of my usual tests. Starting with a single oscillator, I found that, for the most part, the two could sound all but identical. The exception to this was with the maximum deviation of the pulse width; there was a difference here, but only because the range on my Jupe is a bit wider. Limiting this to around 90 percent made the two synths all but indistinguishable.

Next, I added the second oscillator, with and without sync. Inevitably, the differences were a tad more apparent, but if somebody was able to jump into the *SOS* time machine (still hidden in a basement lavvie with a sign on the door saying 'Beware Of The Shark') and retrieve a Jupiter-8 straight off the production line, I suspect that the differences between this and my 40-year-old instrument would be of the same order.

Moving on to the filter, the two synths were again as close as one might hope. (Remember, the ISE-NIN is 'inspired by' rather than an attempt at a clone.) I had to accommodate slight differences in parameter values to obtain near-identical sounds, but it would be facile to expect to recreate a patch on one by putting the sliders in the same positions on the other. With a bit of care, I was able to create filtered sounds that were again close to identical. To be fair, small differences could start to creep in when sweeping sounds with lots of resonance but, at the risk of being excommunicated by the Ancient Brotherhood Of Vintage Is Always Best And Modern Is At Best 'Meh', I'm happy to state that there was nothing here that concerned me.

I carried on testing the rest of the architecture while limiting the ISE-NIN to the features that the two synths have in common, and it became clear that the chaps at Black Corporation have done an excellent job of recreating the soul of the Jupiter-8. Just one significant difference stood out. The digitally generated contours can create a 'pop' at minimal attack and release settings. The way around this is to extend both phases a little; not to the point that they sound sluggish, but enough to eliminate the unwanted sound. I didn't have a problem with this, but it's important that you can find a value that is neither too slow for the patch that you're creating, nor so fast that you obtain the unwanted transient.

So now it was time to try some real sounds. I started by comparing Black Corporation's recreations of the original's factory patches with the same sounds in my Jupiter-8. Some were remarkably close, others could be tweaked to be close, while others captured the spirit of the original patches if not the precise sound. I then recreated some of my own Jupiter-8 patches on the ISE-NIN, and none of the differences would have stopped me from replacing one with the other.

At this point, it was time to stop constraining the new synth by the limitations of the old, and start to invoke things such as velocity sensitivity, aftertouch, poly-aftertouch and MPE. The results could be superb. I was delighted to find that its response curves can be saved on a per-patch basis, and that the ISE-NIN offers several other parameters that allow you to fine-tune how it responds to incoming MIDI messages. A healthy selection of CCs is provided so, in addition to controlling the sound in creative ways while playing, remote programming and automation is a doddle too.

Inevitably, there are some niggles. For example, whenever changing sound banks, the patch number reverts to '1' so, in the absence of a numeric keypad or a search function, you have to scroll to the sound you want. In addition, you'll

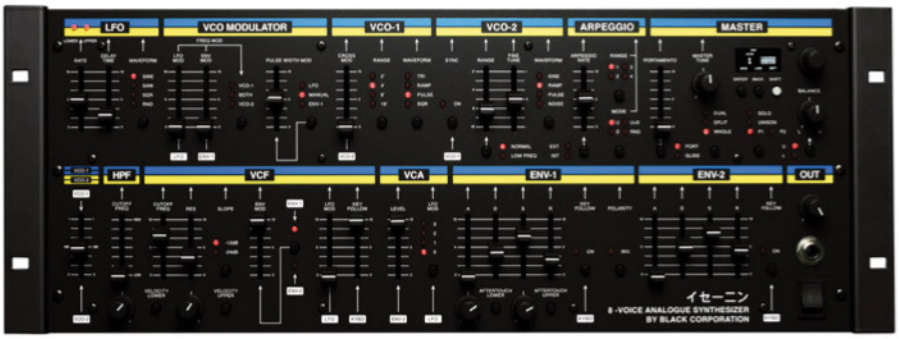
Updating The ISE-NIN Firmware

One of the things that I tend to do when receiving a review unit is check the OS revision and, if appropriate, update it to the latest version. But on this occasion, I didn't do so...

To update the ISE-NIN, you have to download a third-party application called STM32Cube Programmer, which requires that you register at a company called STMicroelectronics. I did so on my MacBook Pro and, having downloaded the software, tried to run it, whereupon macOS said that "SetupSTM32CubeProgrammer-2.14.0 cannot be opened because the developer cannot be verified". This was not a good start. If you decide to overrule Apple and run the program, the manual admits that things may not go smoothly on an Apple Silicon Mac, and then explains how to invoke the Terminal program to get things going. If you succeed, you then power up the ISE-NIN in something called DFU mode and update the synth with two downloaded HEX files that comprise the operating system. The manual warns that the "STM32Cube Programmer may occasionally crash during operation" but, thankfully, this doesn't trash the synth; you can power cycle and attempt the upgrade again. Given all of the caveats (which, I suspect, reveal the synth's origin as a kit for people who really know what they're doing) I decided to leave well alone and tested the ISE-NIN with the version installed when it arrived.

ISE-NIN Ukrainian Version

In addition to the standard ISE-NINs with their Roland-inspired livery, Black Corporation have created five special edition units that sport the national colours of Ukraine on their control panels. The revenue from these will contribute to a fund aiming to buy ambulances and supplies for the civilians of that war-torn country. This is an excellent initiative, and I applaud the company loudly for it.



need a crib sheet to know when you've found it because (like the original) there are no patch names. Given all of the other additions in the ISE-NIN, I think that it would have been more than acceptable to add these too. But the thing that I found most frustrating was the methods for jumping between layers (patches) and Patch Presets, and for saving and recalling them. Everything works, but this is an area that would benefit from a larger screen and a clearer system.

There seems to have been some confusion regarding the number of user memories available, with the manual and the website (as well as various press sites) offering conflicting information. It turns out that the manual is correct. In addition to recreations of the original 64 factory patches found in the Jupiter-8, there are 500 user memories for layers (single patches), plus two banks of 128 user-memories for Patch Presets (Whole, Split and Dual sounds) in addition to 302 permanent factory Patch Presets. If these prove to be insufficient, you can use SysEx to back up and restore banks, although the mechanism for doing so is arcane — you have to press four buttons simultaneously to initiate a transfer — so I don't know why this hasn't been implemented as part of the menus.

The God For You?

If there's one of my vintage polysynths that seems to be liberated from storage more than any other, it's the Jupiter-8. Is this because I can't resist layering its sweet tones across every track? I'm afraid not. It's because manufacturers keep emulating it, and I find myself making the same, detailed comparisons for the umpteenth time. If I'm honest, I think that the software versions are now so good that there's little point setting up the original when recording. Sure, a Jupe looks and feels fantastic, and there are always ways that you can make it sound a little different from any given emulation but, as I've written before, if you're spending half your life hunting for obscure reasons to decry modern recreations of vintage synths, you're not spending enough time making music. Indeed, if you were to record the same track four times with contributions from a Jupiter-8 or Arturia's Jup-8 V4 or the appropriate model in a Jupiter-X or an ISE-NIN, I doubt that your audience would be able to tell the difference. And even if they could, would it matter? If minuscule differences in a few sounds can ruin your tracks, you're writing the wrong music. So where does that leave the ISE-NIN?

There will, of course, be the usual gaggle of Internet trolls who, without ever putting the two side-by-side, will be happy to proclaim that it sounds nothing like a Jupiter-8. My advice to them would be to get out more and stop smoking the funny stuff. So now, with a huge grin, I'm going to write a sentence that I've been anticipating for more than 20 years... Although it's not always quite as accurate to the original, this analogue polysynth inspired by the [insert name of original 1970s/1980s synthesizer here] sounds just as good as the most recent digital emulations. Oh, come on... you've got to chuckle. So the issue with the ISE-NIN isn't the accuracy with which it recreates the Jupiter-8 sound and experience (both of which can be excellent) but its price, which places it in the realms of aspiration rather than affordability.

Actually, there are two issues that I think we should discuss simultaneously because they are related; the cost and the perceived build quality. At around £4500 in the UK, the ISE-NIN is far from cheap and, even in these dark days of synths costing four to five grand, this is an anoxic price for a module. I fully understand the constraints of boutique manufacturing, and there's little doubt in my mind that the price represents the time and effort that has gone into the synth. Nonetheless, it would be easier to overlook it if the ISE-NIN offered premium hardware and a huge touch-sensitive colour screen. But while Black Corporation have balanced the conflicting needs of solidity and lightness, and of small size and sufficient room to get your fingers around the controls, the construction seems just a little too basic, the knobs are of the type that you can buy at your local electronics store, the screen is just a bit too small... and so it goes. While I find it to be attractive and pleasant to use, I have a feeling that many potential customers are going to expect something that looks and feels as expensive as it sounds.

So why would anyone pay this much for a desktop synth inspired by the Jupiter-8? We could argue endlessly about the pros and cons of physical instruments versus software, but I suspect that it's something simpler than that; the original is all but unobtainable and, when one appears, the price is prohibitive. I saw one for sale recently with an asking price of £25,000, so £4500 could be seen to be a bit of a bargain, especially since the ISE-NIN is considerably more flexible than the original. (Even the additional capabilities of the MKS-80 and MPG-80 pale in comparison.) Programming my favourite Jupiter-8 patches on the ISE-NIN, adding lots of performance control, and then playing the results using a Roli Seaboard Rise 2 gave new life to sounds that I've loved since the 1980s, and I have little doubt that there will be some potential owners who view it as both desirable and reasonable value. Others will, of course, deem it to be an outlandish extravagance.

£ £4699 including VAT.
W www.black-corporation.com

DAVID STEWART

Songwriter & Producer

Since breaking through with BTS's 'Dynamite', David Stewart has been making his mark on modern pop music with hit after hit.

SONAL D'SILVA

During the first Covid lockdown in 2020, while some of us were working on our breadmaking skills, songwriter and producer David Stewart was in a bedroom at his parents' home in London. He was working, over Zoom, with frequent collaborator Jessica Agombar, on a track to be sent for consideration by K-pop band BTS, who were looking for their first English-language single.

BTS were already an international phenomenon. They didn't have fans, they had the Army: a legion of diehards who kept track of all things BTS. If they embraced a song, it would be career-changing for all involved. Naturally, every songwriter in town was going for the gig, as Stewart was well aware, but his years in the trenches meant it didn't faze him. "Jess and I had been working for six years consistently before that, writing four times a week, every single week, so it was just another day at the office. We didn't put pressure on it; we never thought 'Oh, we're writing something for BTS and it's going to change our lives! What are we going to do?' That's why I think it ended up working, ultimately, because there was such a free-flowing, normal, easy thing to it. I will forever be grateful for that song."

The turnaround time from writing to approval to release was lightning fast. Released in August 2020, 'Dynamite' broke every record there was to be broken. All the modern pop metrics — streams, views, plays — hit the billions. It spawned a genre of dance covers, where elaborately styled fans performed choreographed routines to the song. George Clooney did a reading of the lyrics for *W* magazine, playing along with one of the year's biggest pop culture moments and echoing what everyone was feeling: it's been a rough year, but this is a real moment of fun. Not bad for a track created in a bedroom, on the simplest possible setup. "It was just a laptop with Pro Tools, a set of ADAM Audio monitors, and a UAD Apollo Twin interface — that's it."

Attention Seeking

Crafting a good pop song is hard enough. Consistently crafting big, impactful, uptempo bangers like Stewart has done is a different matter altogether. As songwriters will tell you, if they're being truly candid, reaching for those minor chords and riding the swell of emotion to a crescendo can feel easier. When asked if he's always gravitated towards the kinds of songs he's become known for, Stewart puts it down to his early years in music. He started out as a live drummer, playing with a band that supported Simply Red on tour across the UK. He then joined the British rapper Example on tour as a guitarist, playing over 600 shows with him and getting a first-hand look at what it took to keep the attention of a crowd of any size. He recalls, "Example was playing to 30 people at first, and eventually headlining arenas, so I got to see an artist's rise to superstardom, and during that process I witnessed what it's like to be in the best live act in the country at the time. No-one

David Stewart in the bedroom studio where BTS's mega-hit 'Dynamite' was written and produced.

could follow us, because from the word 'go' it was just energy, energy, energy."

Going even further back, the seeds of keeping an audience engaged were planted when, as a young boy, Stewart watched his father, the Scottish entertainer Allan Stewart, perform. He recalls, "Everything was based on 'How do we graph an act? How does it [*keep rising*] rather than staying still?' So the ebbs and the flows of a show have always been something that's piqued my interest and [*is reflected in*] the way I make music: it's very impactful, uptempo, fun, and joyous, and it's going to evoke some kind of emotion out of you, whether you like it or not. It's what makes me excited.

"I love chapters in music. I'm a little bored of very linear music that doesn't grow. I want each chorus to grow, each new section to grow. I've learned this from playing live and watching a lot of shows, and I like to embed it into my style of production. The big, in-your-face, impactful, tempo-driven record is naturally what I do and the more I do it, the more I start to understand what that is. By that last chorus, I just want your face to be melting, because everything's happening at that time!"

Another pivotal experience for Stewart was the two years he spent in Atlanta as part of a development deal, working with Ludacris' team. "It was the most incredible experience because I really stuck out like a sore thumb. I went over there as an R&B artist at the time and got to work with some incredible people who came from totally different walks of life than me, who had a whole different perspective on things. I'm used to getting up early, going to the gym at six o'clock in the morning and being at the desk by nine o'clock. Our studio time in Atlanta started at 10 o'clock at night sometimes! You have to acclimatise to the situation you're in and work that room rather than going, 'Well, this is what I do, and that's what I do.' I think it was really important that I succumbed to the process; it's a much more aggressive mentality out there and I loved it. I learned so much about vocal production and it was just such an amazing experience for me. Sometimes I was lucky enough to be able to stay in hotels but I often slept on people's couches; it was my time to really learn the ropes and understand what it was like to be a musician on the grind."

David Stewart: "You have to think of yourself as a business, not just as a musician."

The artist deal didn't work out, but with all that experience under his belt, Stewart set about writing and producing songs, incorporating a variety of techniques he has since become known for: big choruses, ear candy, live instruments, rhythmic vocals, and even key changes, which largely disappeared from pop music in the '90s. He approaches songwriting like an athlete, putting in the hours with the active work of songwriting as well as setting aside time regularly to listen to contemporary music. "I do a lot of listening. You want to keep up to date with what's going on, things like slang in pop culture, or what people are saying now. I don't really lean into that kind of stuff, but it's good to have a grasp of it, to know that we can reference things that are modern, or culturally appropriate. Like anything, it's about doing it over and over again; doing your reps is always going to be the thing that makes you better at it."

Play To Win

David Stewart is a multi-instrumentalist, and guitar in particular features prominently in his productions, "My prized possession is a 1963 Gibson 330TD; it was my dad's 13th birthday present and he passed it down to me. It's beautiful, it just sings, and it's something I use a lot. I've also got a beautiful old relic Tele, that's all beaten up and gold and is insanely heavy, with a really thick neck — it's great for rhythm stuff. Then there's my standard American Strat with Lindy Fralin pickups, which is also really good for blending: you can blend different pickups together to get quite unique sounds through that. I also have a 1963 Gibson LG-2, which is incredible for picking. I have not changed the strings on it because it just sounds so crusty and old and that's how you want it to feel and sound." »

David Stewart's current musical home is this well-equipped studio in Los Angeles.

» His preferred method of recording electric guitars is straight into NI's Guitar Rig. "Never fails. I just keep it simple. I've just moved into this new studio and I'm going to get a nice old Fender Princeton or something to record through because you can't beat feeling the air through the microphones. With Guitar Rig it sounds great, it's very easy to control and change sounds, so that's the reason why I go through that, but if I just want a classic great Strat sound, that I know I won't want to change, then putting it through an actual amp is great — you can really feel the presence of the amp."

To increase the impact of his live instrument recordings, he often layers them with synths or samples. In the chorus for 'Dynamite', Stewart layered the live bass with a Moog bass, making the sound brighter and helping it cut through the arrangement. He uses a similar technique for drums, if the production demands it: "I record my drums and then I will reinforce the kick and the snare with samples using something like a Steven Slate Trigger on a duplicate track. I have a 1963 Ludwig Classic Maple Gold Sparkle with a few different snare drums, and I have everything tuned quite low and thuddy so that when I put my samples on, they blend a little easier than if the kit was tuned high and tight."

Shop Front

Although working from home didn't stop him having a global hit with 'Dynamite', David Stewart now his own studio in LA, which he loves. "This studio is beautiful, it has loads of natural light and I thrive off that. I would really struggle to go into a dark studio every day; I get up early and like to make the most of the day. I've got a grand piano and a drum kit — all the things that I would have loved to have had in my parents' house but I don't think they would have been too happy about it, because I'd have kept them up all night! In terms of synths, I've got a [*Roland*] Juno 106, a Moog, and a Yamaha SK30 which is a really interesting old-sounding '80s synth that Mike Dean uses a lot. It's really cool. Sat next to my drum kit, I've got 12 Neve 1081 preamps and eight API 3124Vs. For the grand piano, I have some Neve 4081 quad mic preamps, so it's different flavours that you can just dial in basically."

Moving to LA has also allowed Stewart to take his career to the next level. "As far as the pool of artists I had access to, I think I hit my ceiling in London. The whole thing with LA is, you get a call on a Thursday night saying, 'Can you come and meet the artist on Friday morning?' When I was in London, I would say, 'I can meet the artist, but I'll be in LA in three weeks time.' By the time you get there, the opportunity is gone and someone else has got the job.

"Everything I've done to this point has been like a three-pointer, like trying to shoot from the halfway line, and I've been good at that because I'm good at placing records and writing songs that resonate with people — but the way to really go to the moon is to be with the artist. Before this point, I was running around with my laptop in LA and just doing sporadic sessions, but now that I have my studio, I have a shop front. I have a hub with my own taste in it, with the way I like to decorate things — people are coming into my world. Now that I'm here and I have the opportunity to be around some great artists and other songwriters at that top level, I feel I can go to the moon."

Mind Over Matter

David Stewart had been preparing for his overnight success for over a decade when it finally happened. What his years

A multi-instrumentalist, David Stewart likes to have his drum kit permanently miked and ready to go.

in the background gave him was clarity about what it takes to survive in the industry. To hear him tell it, 'making it' is as much about getting your head in order as it is about strengthening your music chops. "As an artist coming up, music is a tiny bit of the equation. It's awful, but if you have resilience, if you can be strong, if you can get up after you've heard 'no' a gazillion times, that's the most important thing.

"I would say to a new artist, make the word 'no' mean nothing to you. As soon as it means nothing to you, you're on your way, because you are going to hear that word for the rest of your life, no matter how successful you are. Before things happen for you, everything is a 'no' and then you get that one 'yes', and you're off to the moon. For 10 years everything was 'no' for me. I used to play shows and sometimes my dad and the sound man were the only people in the audience. It was so demoralising, but after a certain point, you go, 'Well, that's just what it is.' This is the music industry, I signed up for it. And then suddenly some God goes, 'You know what, you've taken eight thousand nos, so I'm going to give you a yes because you really deserve it.' I really, truly believe that it's about how long you can stand up before you get your 'yes'. That's what separates the winners from the losers."

As well as being able to withstand rejection, Stewart highlights another key piece of the puzzle that is crucial to success, one that creative people tend to hate: the business side of things. "The music business is equally important as the music-making process. Take every meeting; you don't know what's going to lead you to something, you don't know what you're going to learn."

The contemporary pop market is particularly crowded. Even landing a credit on a hit single is not enough to guarantee a good payday, given that more often than not, you're sharing that credit with a laundry list of other writers. Everyone gets a smaller piece of the

»

» pie, which has already shrunk in size due to the other upheavals in the music business. "We are in a time where it's very difficult to make money off music unless you're really in that top end of writers and producers, and even then, I know hit songwriters and producers who have a second job because [*of how difficult the business is*]. So you have to be aware of your surroundings and you have to think of yourself as a business, not just as a musician."

While he was paying his dues, he was also building self-sufficiency. "The reason why I got into production so much was because I didn't want to go elsewhere to get what I needed. I had the song and then I went, 'Well, I don't want to have to wait to get someone to produce it,' so I taught myself how to do it. I don't want to wait for a guitarist, I'll do it; I don't want to wait for that drummer, I don't want to wait for that artwork. I'll just figure out a way to do it. Learn how to do every aspect of this business. It doesn't mean you have to be great at everything, but know what you want, or, even more importantly, know what you don't want — that sets you up to be in a place where you become indispensable. You almost need to be a jack of all trades now, just doing 100 percent."

Random Walks

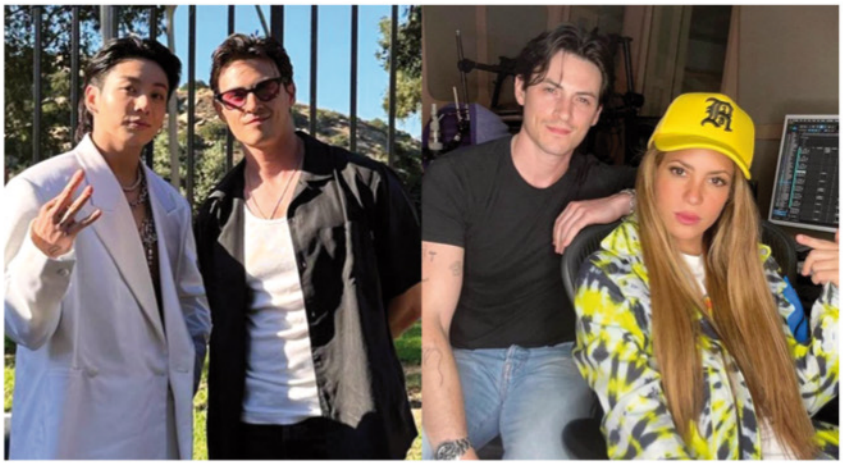
Pop music has become surprisingly varied and unpredictable, and it's hard to spot trends, or pinpoint sounds that define contemporary pop in the way that gated snare reverb did the music of the '80s, or rock & roll guitar that of the '50s. "I think we're in a time where I actually can say there is no [*one*] sound that

Vocals Are Everything

The lead vocal is the undisputed focal point of a pop song, and David Stewart is mindful of that from the beginning. "I almost never, ever start with a beat any more. I've changed my process in the last couple of years to making disgustingly basic demos during the initial process of the song, because if the song sounds great on a piano or a guitar with a horrible drum loop under it, I know it's going to translate into something that feels even better when we put real production behind it. I [***record***] the demo vocal and work backwards from there, because the vocal is king, or queen, and we need to make sure that's the most important thing."

His mic choices and vocal chain are guided by the voice he's working with, he explains, "It's different for different artists. I've got a Sony C800G mic which is great for pop and rap because it's a bit more punchy, bright, and in your face. I've got an old '63 AKG C12 which I used with a new artist called Hana Effron that I'm working with; she has a rich bottom end in her voice and the C12 really captures that bottom end and is also very silky on top. Next, I go into a Neve 1073 preamp and then a Universal Audio 1176, Tube-Tech CL-1B, or an LA-2A compressor, depending on what we're doing. With Shania [*Twain*], we used a Telefunken ELA M 251E, which is just an amazing all-round microphone; I actually think it sounds great on male vocals as well. I did the whole Olly Murs album *Marry Me* using the 251 going into the 1073 and the LA-2A. Shakira wanted to work on an SM58, even though I offered her a plethora of expensive microphones and all the fancy gear. So it's really about tailoring it to every artist, it's not one size fits all."

Once the vocal has been recorded, Stewart typically adds basic effects in the box: "It's very simple stuff, nothing crazy. For reverb, you can't go wrong with a plug-in like Valhalla, Avid ReVibe, or Waves R-Verb. I also use the Waves H-Delay plug-in, with a quarter-note, eighth-note, and slap delay set up — that's a template of sorts, so I can dial in what's needed. Then, most of the time, I'll run my lead vocal through something like a Waves CLA Vocals plug-in so that it comes out feeling a little wider — I don't put anything on it apart from the spreader or the doubler. When you listen to a mix by someone like Serban Ghenea, there is a wideness to his lead vocals; everyone in the world is trying to find out how he does it. This is as close as I can get to it before I actually go to Serban and say, 'Can you just do your thing to it?'"



One of the most important aspects of pop production is to get the best possible vocal performance from the artist, whether that's K-pop star Jungkook (left) or Shakira.

What makes a pop vocal really land, though, are the subtle performance qualities that are as much about feel as they are about sound, and which the producer tries to coax from his stars. "I often find when you're producing vocals, as with every instrument, the space is what creates the rhythm. It's not just what you're playing, it's the actual space that you leave that creates the groove, and it's the same with vocals. If you want a vocal [*with a strong attack*], it's not just about the front of the line, it's often the back end of the line. How quickly you come off something is where you find your pace in the vocal; if you're elongating that last word, it will make the rest of the line feel a bit sleepy. Then, it might be about adding in a breath or adding [*a vocal element*] that just picks up on the rhythm very subtly and will dance with the rest of the instruments. Sometimes you don't even want to hear it, you just want to feel it's there. Michael Jackson was the king of that; it's really where it comes from, and I do use it in a lot of my songs."

Stewart also highlights big background vocals as a key element of his production style. Whether it's for BTS or the artists he's worked with since — including Shania Twain, Kygo and BTS member Jungkook, with whom he scored another number one hit, '3D' — Stewart performs all the backing vocals himself, and shares two techniques that make them effective, "I think the important thing is to mimic the lead vocal. You almost need to become an impressionist of the way [*the **lead***] vocal has been sung; don't just do what comes naturally to you. It's different for different people, and some voices blend much easier. The Jonas Brothers, Jungkook, BTS and I have similar tonal qualities in our voices, but with someone like Shania it's different, so I have to mimic her. It gives a rich sonic quality to the song. I might also ask an artist to change a vowel sound because it will come across differently over the microphone than it would in the room. What I would then do, when it comes to background vocals, is mimic the same kind of mouth position for the vowel."

Then there's the technical aspect of it: "You have to do things [*to make the backing **vocals blend***] like taking out all of the 's' sounds at the ends of words, so you're not getting eight different esses [*stacked on each other*], you're just getting your lead vocal."

The Yamaha grand piano at David Stewart's LA studio is likewise ready to be recorded at a moment's notice.

works now," agrees Stewart. "It's totally fascinating. In the mid-2000s you'd hear a song and you'd go, 'That sounds like a hit,' and it would usually be a smash hit, whether it be a Katy Perry song, or any of those big records from that period of time. If something sounded enormous, it would usually become enormous. Now, we're not in that position at all. The weirdest, janky, funny little song from the back end of nowhere trends on TikTok and becomes a number one hit, and the song that we all think is going to be the biggest thing in the world — that is a singalong and sounds huge and has everything that in 2010 would have made it a number one for eight weeks running — doesn't get a look in. It's exciting. We can now do stuff that doesn't necessarily fit the confines of what we remember pop music as being, because pop music is everything now. There is a space for every type of artist to be able to shine because everything is so big; there is a much wider scale of places to listen to everything from.

"I think it's down to the fact that we're in this streaming culture and social media culture that drives the culture of music. There is no one funnelled place to listen to or consume music anymore like it used to be with radio. Radio chose the hits back in the day; there were only 40 songs on that playlist and you were force-fed a song until you liked it. Now, we have 100,000 songs coming out every Friday, and it's down to what trends or gets a moment."

As a songwriter and producer, he responds to the randomness of it all by relying on his instincts: "Always follow your gut. Don't try and chase a trend, because if you're chasing a trend, you're often behind it already. I would say [*the current landscape*] is even more of a reason to just do what feels right to you. Lean into what feels right, because there is no tried and true any more, it doesn't exist. A small janky little ukulele song with a trap beat could be a number one hit. Everything I've done has always felt right to me; whether it becomes a hit or not, it doesn't matter. I'm always proud of it, and if it becomes a hit, and people resonate with it, that's amazing. There is no rhyme or reason for what's working, and you would hope the good stuff rises to the top but if it doesn't, then so be it. All you can do is do what feels right to you."

Strong Words

Pop lyrics get a bad rap for being nonsensical or trivial, but as David Stewart points out, that doesn't mean they aren't important. Successful pop lyrics flow satisfyingly off the tongue; their purpose is to feel good to sing. There's a real craft to it, and it's harder than it looks. "I've been fortunate enough, over the past few years, to be able to work with people like Max Martin, Rami Yacoub and ILYA. They are almost pre-programmed to go with lyrics that are phonetically pleasing to your ear; they take real pride in that Swedish style of linear, almost mathematical, writing. I naturally used to do that anyway, because I'm good at lyrics, but it's not my best; it's not something that comes as easily to me as the rest [*of it*]. I would much rather work with someone who is a great lyricist, but I do know when something sounds right."

'Dynamite' provides a great example of how a minor modification can have a major influence over the feel of a song. "So call me Mr Dynamite" was changed to "Light it up like dynamite", the internal rhyme in the second version making it roll off the tongue better. Stewart and Agombar, with whom he writes frequently, have their own lingo for lyrics that jump out at them: "We call them zingy, sparkly lyrics — things that make your ears kind of tickle a little bit. For example, something like 'Shine bright like a diamond' [*from Rihanna's 'Diamonds', written by Sia, Benny Blanco and Stargate*] has that zingy thing to it. We call them shiny words: things that catch your ear, that rhythmically make sense, even if [*the line*] doesn't always make sense. It serves a purpose. With pop music, if you can get a great lyric, that's everything; however, the phonetic, the actual sound of the word you're using, can be equally as impactful as a great lyric."

Royer Labs FlexBar

Microphone Stand Accessory

The FlexBar can make all sorts of dual-mic placement jobs quick and easy.

HUGH ROBJOHNS

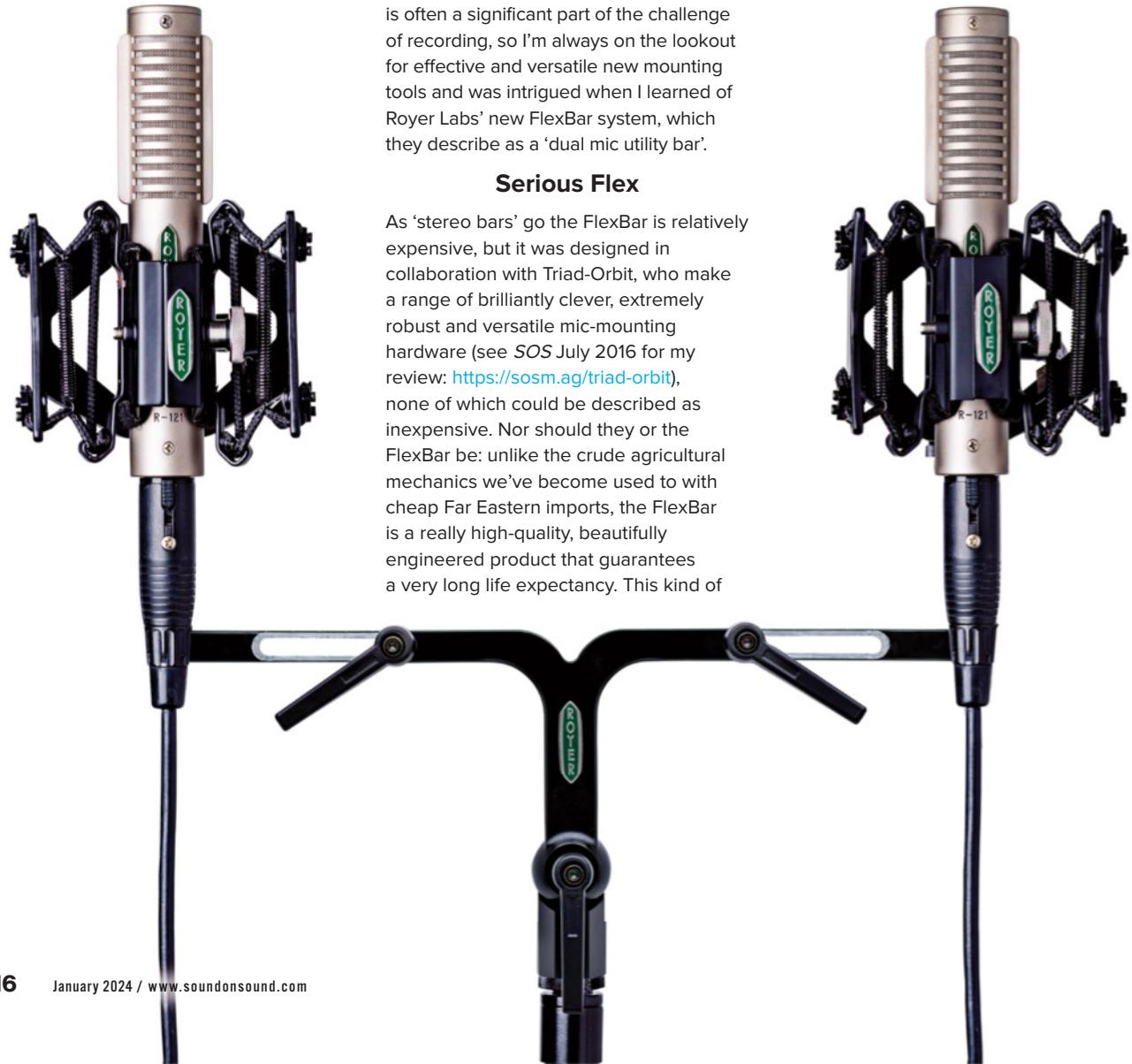
There's a popular adage amongst recording engineers that details in order the things that most affect the quality of a sound recording. It starts with the musical composition, then the musical arrangement, next the performer(s), the recording environment, the mic positions, the mic selections... and finally the rest of the technique and technology: mixing, mic preamps, signal processing, converters, file formats, and all the rest that we tend to get hung up on but which, in the grand scheme, make relatively little difference to the end product.

If you think about it, this list makes perfect sense. If you don't start with a really great song that people want to hear, in a good arrangement and performed sublimely, the best mics in the world won't make it any better. And if you don't place those world-class mics in the right places around the instruments no one will hear the performance as intended either! In my experience, finding ways of getting microphones where you want them to be is often a significant part of the challenge of recording, so I'm always on the lookout for effective and versatile new mounting tools and was intrigued when I learned of Royer Labs' new FlexBar system, which they describe as a 'dual mic utility bar'.

Serious Flex

As 'stereo bars' go the FlexBar is relatively expensive, but it was designed in collaboration with Triad-Orbit, who make a range of brilliantly clever, extremely robust and versatile mic-mounting hardware (see *SOS* July 2016 for my review: https://sosm.ag/triad-orbit), none of which could be described as inexpensive. Nor should they or the FlexBar be: unlike the crude agricultural mechanics we've become used to with cheap Far Eastern imports, the FlexBar is a really high-quality, beautifully engineered product that guarantees a very long life expectancy. This kind of gear is a genuine investment designed to work day in, day out in tough professional environments over a long lifetime.

Most 'stereo bars' are simple designs, comprising a rail with a central stand mounting point and a couple of threaded mic mounts that are usually movable along the bar to set the required distance between mics. The FlexBar takes a simple but very different approach that affords a level of flexibility and configurability I've never found in any mic mounting system before. At its heart is an aluminium T-shaped bar with slotted cross arms and a knuckle joint at the base to fix the system onto a mic stand. The upright of the T is around 130mm high and the cross arms extend about 70mm to each side. The knuckle joint is locked in position by a re-positionable lever arm, the length of which makes tightening very easy, although very little tension is actually required. The mating surfaces of the knuckle clamp have fine radial serrations so that once closed together they are mechanically locked in place across their



It's probably no coincidence that the FlexBar neatly accommodates two Royer ribbon mics in a Blumlein array!

entire surface area. A spring pushes the two interlocking surfaces apart when the lever is loosened, so they can then rotate freely, and a miniature ball race with pressure washers either side ensures the lever's action is smooth and consistent.

Slotted into the T-bar's cross arms are movable L-shaped extensions, terminating with a mic mounting thread and locking ring. In fact, all of the fixing threads on the FlexBar are the American standard 5/8-inch type, but FlexBars destined for export markets (like the UK and Europe) are shipped with a 3/8-inch threaded insert for the stand-mount socket, and a pair of 3/8-inch adaptors for the mic mounts. The L-shaped arms are slotted to allow extension over a 50mm range, with up to 225 degrees of rotation about the hinge point. Locked, again, with long, adjustable lever arms, serrated washers either side of the extension arm grip it securely in place with little tension on the clamp. Indeed, over-tightening will tend to flatten the serrated washers and reduce grip over the long term rather than maintain it!

Along the tops of the extension bars are centimetre marks indicating the (horizontal) mic mount distances from the centre (scaled from 10 to 15 cm), as an aide to quickly configuring the stereo bar. Overall, the horizontal span between mic mounts ranges from 20 to 30 cm.

In Practice

The most fiendishly challenging aspect of the FlexBar is figuring out how to adjust the arms to get the thing back into the foam cutout in its storage box after use! But actually configuring it to support a chosen mic array is obvious and fast and, most importantly, the result is reassuringly sturdy. Taking the simplest configuration first, with the movable arms extending horizontally from the T-bar arms, the FlexBar serves as a reasonably standard-looking fixed stereo bar to support side-address or end-fire mics in all the usual coincident and near-coincident array formats (XY, ORTF, NOS and so on). For end-fire mics where the capsules need to overlay one another, rotating one or both movable arms slightly accommodates that without needing extra thread adaptors to gain the necessary height.

For side-address mics, a neat-looking option is to allow the movable arms to drop downwards and face outwards, so that the mics rise up along the sides of the mount with around a 30-40 cm capsule spacing, depending on mic suspension hardware. This is a practical arrangement for near-spaced stereo pairs of omnis, figure-8s or cardioids in many situations.

Of course, Royer are best known for their ribbon mics, which are inherently side-address figure-8s — a format which is notoriously tricky to mount neatly in a Blumlein array since it requires the mics to be mounted vertically, head to head. The FlexBar accommodates this requirement with ease — it's as if it were designed specifically for that purpose! Simply rotate the knuckle joint so that the T-bar extends horizontally from the mic stand, move the arms to their fullest extent, attach the mics to face into the centre of the bar and rotate them to the desired ±45-degree angles. The result looks elegant, and it's simple to achieve and very solid in practice.

ALTERNATIVES

I am unaware of any mic-mounting system quite this versatile. The **Shure A27M** comes close for many applications, but the **FlexBar** can do things the A27M can't.

Moving on from classic stereo mic array configurations, the FlexBar can also be used to position any two mics of whatever type wherever required. For example, if you want two mics on an acoustic guitar, one near the 12th fret and one nearer the tail, the FlexBar can do that. Need to mic up a singing guitarist, with one mic for vocal and one on the guitar body? The FlexBar can sort that out in a trice. Want both top and bottom mics on a snare drum? The FlexBar can do that easily. Ditto, for a pair of mics dropped through the open lid of an upright piano, or even close mics near the hammers tucked under the lid of a grand piano. The FlexBar can do everything a standard stereo bar can do, and a lot more besides, as well. Imagination and requirements are the only limits!

Verdict

While handling and adjusting the FlexBar, its solidity and ruggedness are apparent. This thing genuinely oozes quality engineering, and its supremely versatile configurability is quite unlike anything else I've used. If I had to nitpick (I do — it's in my job description!), I'd suggest that, perhaps, the FlexBar could benefit from some built-in clips or clamps to assist with fast and neat cable dressing. But that's very obviously not a deal-breaker, and I got on fine using my usual black Velcro wrap-around ties.

While pricier than traditional stereo bars, the asking price is reasonable. Indeed, looked upon as a tool that will remain in use over decades — it really will get used in almost every recording session — the investment seems a very wise one. All the best designs seem simple and obvious once someone has brought them to market, and the FlexBar is no different. It's a genius product, and once used you'll want to go on using it for everything. I know I do! ■■■

£ £252 including VAT.
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W https://royerlabs.com

Royer Labs FlexBar
£252

PROS
- Solidly and beautifully engineered.
- Supremely versatile and configurable in ways no other utility bar can match.
- 3/8-inch adaptors included in export sets.

CONS
- Cost.
- No built-in cable-dressing clips.

SUMMARY
An ingenious but simple and elegant solution to myriad awkward mic-mounting conundra.

Vienna Symphonic Library Synchron Duality Strings

Sample Library

The latest Vienna string orchestra offers a unique dual perspective.

DAVE STEWART

The most recent instalment in an unbroken line of sample products stretching back to 2000, Synchron Duality Strings began life when VSL founder Herb Tucmandl woke up one morning with double vision. I hasten to add no ocular disturbance or medical emergency was involved, and as far as we know the Vienna supremo wasn't suffering from a hangover — this was more a case of a blinding flash of inspiration which marked the genesis of a unique double-whammy project.

Tucmandl explains the concept: "When working with ambient (ie. hall-recorded) libraries, sometimes you want to have a closer, more direct sound. To achieve this, you have to use close mics as you would in real world orchestral recordings. When doing this you get the section leader more prominently in your mix, more or less a zoom in on an individual player in the section. In a perfect world you would have moved the whole section closer, which is not possible in a convenient way."

Rather than addressing the problem with different mic mixes or layering separate performances, Tucmandl proposed a radical solution: "For a perfect result you would need a performance of all musicians in a wet and a dry environment together at the same time. Thinking about the options at the Vienna Synchron Stage, I realised that it is possible. We have two stages, our famous big hall and the smaller dry hall. Doing film scoring for seven years now, our musicians are trained to perform with headphones to deliver perfect results. So what we did was sample two string orchestras in two stages at the same time."

In musical terms, this equates to a simultaneous recording of a symphonic string section playing in the huge Synchron Stage A and a smaller chamber ensemble positioned in the smaller, dryer-sounding Stage B (see box below for section sizes). The musicians

VSL Synchron Duality Strings

From €595

PROS

- Offers simultaneously recorded symphonic and medium-sized string ensembles in one package.
- Contains VSL's usual comprehensive range of articulations and creamy legatos.
- You can instantly switch from a cinematic ambience to a studio acoustic, or layer the two for the best of both worlds!
- The players turn in superbly precise and unified musical performances.

CONS

- VSL pro-level libraries aren't cheap, and never have been...
- ...but you can save a lot of money by not buying the extra mic positions.
- The elaborate articulation switching system might confuse some beginners.

SUMMARY

Fancy two string orchestras for the price of one? Synchron Duality Strings features a simultaneously recorded 52-piece symphonic string section and a smaller chamber ensemble playing in separate rooms, so you can use them separately or in combination. Both groups feature first and second violins, violas, cellos and basses performing a large menu of precisely executed articulations, with a host of microphone position available for the symphonic section. A unique and versatile string library which will work equally well for cinematic, classical and pop productions.

played to a click with a video link to the conductor and their colleagues in the adjoining room, acoustically isolated yet perfectly synchronised.

Overview

The string players are all members of the Synchron Stage Orchestra, a hand-picked ensemble drawn from the top Viennese orchestras. Looking at the photos, it's obvious that the smaller Synchron Stage B space will create a more intimate, dryer studio-style ambience, in contrast to the big reverberant sound of Stage A. Rather than trying to replicate the intensive multi-miking used in the latter, VSL's Chief Recording Engineer Bernd Mazagg devised two different setups inside the dry stage, using different mic types to create a bright and dark option.

Synchron libraries run exclusively on the dedicated Vienna Synchron Player, which works as a plug-in and standalone on Mac and Windows systems. 16GB of RAM and an SSD drive are recommended. VSL have now adopted the iLok licence management system, which can be used on the iLok Cloud or with a physical USB key. The latter requires an iLok 2 or 3 key (the old iLok 1 won't work — boo!) but comes with the advantage of excellent free instruments (hooray!). You can read all about it at the company's www.vsl.co.at/en/News/Hello_iLok page.

As with other VSL Synchron products, the Standard Library and Full Library include the same articulations and four basic mic positions, but the latter, more expensive option adds four extra mic perspectives (Ribbon mic, Main Surround Stereo, High Stereo and High Surround Stereo), the latter three of which would come into their own in 5.1 and immersive mixes.

Having immediate access to symphonic and chamber string ensembles playing simultaneously in radically different acoustics opens up a wealth of creative possibilities. You can use the two sections in combination, or separately to provide contrast. When layered, it's notable how the smaller ensemble adds definition, clarity and overall brightness to the symphonic section, revealing more detail, enhancing bow attacks and increasing the rhythmic drive of the spiccato and staccato short notes. The brightening effect of the chamber ensemble is evident

» regardless of which mic set up you use, and serves as proof of concept for Herb Tucmandl's vision.

All Together Now

Synchron Duality Strings' presets group together all the articulation patches of each individual section, thus, you can access all of the first violins' playing styles within one preset, and the same is true of the second violins, violas, cellos and double basses. Presets load with both the Stage A and Stage B samples in place, after which you can use the Synchron Player's mixer channels to alter the balance and timbre of the two groups.

There are four preset types: the old-school 'Velocity' assigns dynamics to keystroke velocity, while 'VelXF' and 'VelXF sus' presets allow you to crossfade between the instruments' dynamic layers with the mod wheel (a very effective way of creating crescendos, diminuendos and subtle volume swells). A new 'VelXF + Velocity control' category also uses the mod wheel but adds key velocity to determine long notes' attack behaviour, and occasionally to velocity-switch between different playing styles (for example, spiccato and staccato).

Those versed in the dark arts of MIDI orchestration will no doubt dive straight into the individual sections and start furiously programming viola parts, but if you're simply looking for a quick, playable full strings sound for sketching or playing live, there are some magnificent 'tutti' presets which map out the five sections over six octaves with a full set of playing styles.

I had an indecent amount of fun playing these presets, emerging after a few hours with a good understanding of what the Synchron Duality strings are capable of. Played in a choice of light and strong ('molto') vibrato styles, the full section's long notes sound majestic. I generally preferred the lighter vibrato, but you can crossfade between the two and enjoy a seriously lush sound when they overlap.

If your mission in life is the creation of rhythmic string ostinatos, you'll be pleased with these strings' spiccato short notes. Played in normal and tight styles (the latter featuring more round robins), they are tremendously strong and propulsive and have a sumptuous full-bodied sound. The pizzicatos are equally impressive: tight,

The Synchron Player GUI's mix tab. Samples are mapped to the white keys, coloured keys are switches, which select the basic articulation, articulation style, vibrato style, release and attack options.

Synchron Duality Strings' pre-recorded runs play in all 12 keys starting on every note of the scale.

plump-sounding and melodious, they're complemented by some brusque, tidily played col legno hits. They also have an evil twin in the form of an utterly vicious 'snap pizzicato' delivery, which when played chordally sounds like a gang of brigands thrashing a wooden beer barrel with bunches of birch twigs.

Artic Conditions

An altogether more subtle short-note bowed articulation is the détaché. Its maximum length is about one second, but as Herb Tucmandl points out, you can play this style shorter with very realistic results, and the provision of soft, regular and marcato attack options increases its flexibility. Speaking of note attacks, this collection lacks the traditional Vienna 'sforzatissimo' dynamic delivery, which I would always turn to in moments of unbridled musical passion. This library uses the less forceful sforzato style as a sustained note attack option, marking a retreat into slightly calmer orchestral waters. »

SOUND ON SOUND

VIDEO

VIDEO DOCUMENTARY ORIGINALS

IN ASSOCIATION WITH

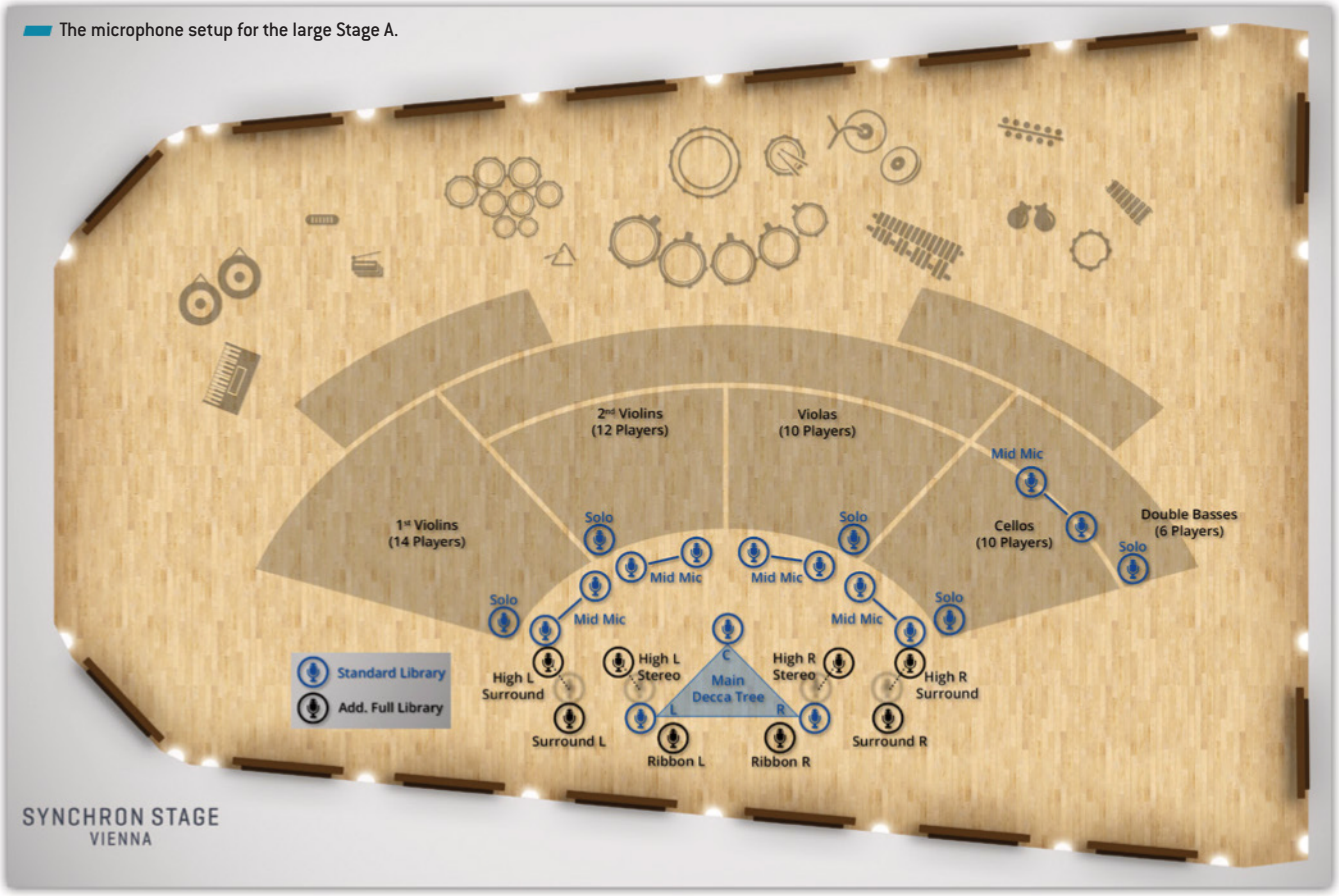
UNIVERSAL AUDIO

FRASER T SMITH PRODUCING 'SET FIRE TO THE RAIN'

'Set Fire To The Rain' was Adele's third consecutive US number one, confirming her status as a superstar. In our exclusive video feature, Fraser T Smith tells us how he and Adele wrote the ultimate power ballad, breaks down his production and explains why a humble Yamaha upright piano was the best investment he ever made.

www.youtube.com/soundonsoundvideo

The microphone setup for the large Stage A.



When it comes to fashioning melody lines, VSL's legato patches have always yielded superior results, and that still holds true. The 'agile' legato style works beautifully across all five sections, transforming everything I played into a smooth, graceful melodic continuum. Normally I'd expect to have to tidy up the MIDI notes after programming a fast legato line, but that wasn't necessary here — the agile legato patches always seemed to know what to do, and kept pace with the fastest lines, trills and flourishes I'm physically able to play.

In addition to the normal and agile legato modes is a nice slurred legato which introduces subtle expressive slides between notes, and a fabulous portamento variant which exaggerates and elongates the slides. The latter is the style you need for the exuberant, sinuous 'Bollywood strings' sound heard in Indian film soundtracks and pop records. Superbly played by all five sections, it's one of the highlights of the library.

You can also use the library's excellent glissando samples to program dramatic-sounding slides which end on a sustained target note. Chromatically mapped and played up and down, fast and slow over an octave, fifth and major second ranges, they're a great source of cartoon-style comedy effects and so-called 'disco slides', while the major second variant can add evocative and subtle grace note slides to a melody.

Digging deeper into the artics, one notices the VSL's trademark 'bold' and 'agile' variations cropping up repeatedly. Though often hard to tell apart, the difference between these two lies in the speed of transition between notes. Herb Tucmandl explains, "The agile version becomes important when performing melodic phrases. The normal (bold) version offers a kind of starting sound for each new note — in most cases musicians wouldn't play a phrase that way, they would perform it in a way similar to the agile style, which has a more natural fluidity." So as a rule of thumb, you should use the 'agile' style for faster melodic phrases.

Runs, Trems & Trills

Though you can create ultra-realistic runs with the agile legato artic, pre-recorded runs offer a quick and convenient alternative. In this library they're played legato and détaché in a straight major scale, up and down in all 12 keys in a choice of octave, fifth and major third ranges. The runs start on every note of the scale, so if you select the key of G and choose the ascending version, playing a note of G triggers a G-A-B-C-D-E-F#-G run, a note of A triggers A-B-C-D-E-F#-G-A, and so on.

Although the runs lack whole tone and chromatic scale variants, you can trigger a minor key run by playing a note three semitones down from the root note — in the key of C, that creates the natural minor scale of A-B-C-D-E-F-G-A. However, if you want the more exotic harmonic minor scale (A-B-C-D-E-F-G#-A), you'll have to switch to agile legato mode and play it yourself! If you're having problems with the fingering, remember that you can always temporarily slow your DAW's tempo down to a manageable speed to record a scale run.

Tremolo is the classic nail-biting orchestral strings style denoting tension and imminent danger. Nowadays it sounds a bit corny, but this library's slow tremolo variant sidesteps the clichés and imparts a subtle timbral motion to sustained notes. The so-called 'measured tremolo' artic is actually a set of fast

16th-note repetitions, brilliantly played across the instruments' full ranges in a range of tempos from 120 to 180 bpm — instant, driving rhythmic ostinato material. Another orchestral strings library staple, trills come in the standard semitone and tone flavours, but there are also minor and major third variants which can add an electrifying shimmer to single high-register violin notes.

In Operation

As reported in previous VSL reviews, the Synchron sample player incorporates an ingenious and elaborate switching system which enables you to change playing styles in real time. This can be as obvious as switching from bowed to pizzicato, or as subtle as quickly changing from regular legato to slurred legato in order to add expression to one particular note of a melody. Either way, the colour-coded keyswitching system works brilliantly, and once you get your head around the layout and become familiar with the various articulations, it can be used fluidly and intuitively.

The player's built-in mixer contains a wealth of useful effects, including parametric EQ, filters, compression, expander, delay, reverb, chorus, phaser, tape saturation, power pan and stereo widener. The reverb includes an excellent plate effect which adds a cinematic dimension to the concert hall acoustic. To help you get started with integrating these effects with the numerous microphone channel options, VSL have supplied a large number of mix presets ranging from close, relatively dry perspectives to panoramic ambience-drenched soundscapes. Being something of a reverb addict, I enjoyed the 'Lush Long' and 'Airy' presets, which make good use of the plate and room reverbs.

Conclusion

Doing the maths, even your dull-witted reviewer can see that buying a symphonic and chamber string section in one package is likely to be cheaper than purchasing them separately. But why buy this particular library? The sound quality is stunning, but for me a deciding factor would be the amazingly unified nature of the performances. Having spent seven years recording film scores and classical works together in the same building, this team of players have developed a musical empathy and collective feel which allows them to deliver the most precise, expressive and finely coordinated string samples you're ever likely to encounter.

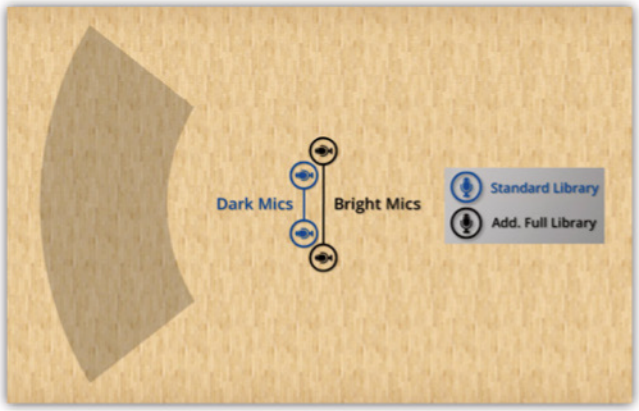
If you're considering the benefits of the Full Library, I can say that in the vast majority of cases it's a luxury you don't need. Gone are the days when you had to shell out extra cash to acquire the full range of VSL articulations; nowadays the only difference between Standard and Full versions is the number of microphone positions, so unless your work routinely involves surround and immersive mixing, the Standard version will do you proud.

It's good to know that having unleashed this innovative 'two ensembles performing simultaneously in different rooms' concept, VSL are not about to abandon it — the company have already announced that con sordino (muted) and smaller ponticello and flautando libraries are in the pipeline, which I'm sure will please avid Vienna collectors. Having heard Synchron Duality Strings bursting into life in my music room, it's clear that Herb Tucmandl's bold vision works supremely well. Duality is now a reality, and I look forward to hearing its future manifestations.

£ Standard Library €595, Full Library €960. Prices include VAT.
W www.vsl.co.at

Mic Positions & Library Sizes

The Standard Library and Full Library (sometimes referred to as the Extended Library) of Synchron Duality Strings are separate downloads. Both include the same articulations, the difference lies in the number of microphone positions. Buyers of the Standard Library can upgrade to the Full Library at any time by purchasing the Extended content, resulting in the same overall outlay as buying the Full version off the bat.



The simpler microphone setup used in the smaller Stage B.

The 'Dark' and 'Bright' mic setups used in Stage B consist of a mix of two microphone pairs — a Royer SF24 stereo ribbon mic and Sennheiser MKH 8040 for the dark option, and a Neumann M149 tube mic and Ehrlund EHR-M for the bright. The Standard Library also includes a RAM-saving Stage A room mix composed of merged samples from the individual positions. For those that need it, adding the Extended Library will expand your sound world into the realms of 5.1 surround, Dolby Atmos and Auro 3D 9.1 immersive formats.

Stage A	Standard Library	Full Library
1. Solo Mic (Mono)	✓	✓
2. Mid Mic (L-R)	✓	✓
3. Ribbon Mic (L-R)		✓
4. Main/Room Mic — Decca Tree Stereo (L-R)	✓	✓
5. Main/Room Mic — Decca Tree Mono (Centre)	✓	✓
6. Main Surround — Stereo (L-R)		✓
7. High Stereo (3D) — Stereo (L-R)		✓
8. High Surround (3D) — Stereo (L-R)		✓

Stage B	Standard Library	Full Library
1. Dark Mics (L-R)	✓	✓
2. Bright Mics (L-R)		✓

Library sizes	Standard Library	Full Library
Sample amount	328,926	603,031
Download file size	51.9GB	101.6GB
Installed file size	121.2GB	242.3GB

Handheld Capacitor Mics

Bring studio sound to the stage with these live-oriented capacitor microphones.

LUKE WOOD

Dynamic microphones are the go-to choice for many live engineers, and for good reason: they generally offer sturdy construction and high SPL handling, meaning they stand up well to the demands of live use. However, the extended high-frequency range and improved transient response of a capacitor-based design can often deliver much more natural-sounding results on many sources, not least vocals. In this month's Spotlight, we take a look at a range of handheld capacitor mics that have been designed first and foremost for use on stage.

AKG C5

AKG say that the C5 will deliver a vocal tone that will cut though the loudest mixes, and also offers a choice of two tonal options thanks to an included presence boost adaptor that can be fitted to the capsule assembly. A solid die-cast body and rugged grille construction help the C5 stand up to use in a live enviroment, and the inner casing has been designed to protect the capsule from corrosion and humidity.

£ £209 including VAT
W www.akg.com/Microphones/Condenser%20Microphones/C5.html?

AKG C7

Another offering from AKG comes in the form of the C7, which features a similar physical design to the C5 but offers a supercardioid polar pattern and improved technical performance. An open-space capsule design reduces air turbulence and inner reflections at the rear of the capsule, which AKG say yields an extremely consistent polar pattern throughout the entire frequency spectrum, resulting in excellent feedback and spill rejection.

£ £235 including VAT
W www.soundonsound.com/reviews/akg-c7
W www.akg.com/Microphones/Condenser%20Microphones/C7.html?

Audio-Technica AT2010

The Audio-Technica AT2010 has been designed specifically for close-miking vocals, and aims to bring the articulation and intelligibility of the company's renowned 20 Series mics to the stage. The mic employs the same 16mm low-mass diaphragm found in the AT2020, offering an extended (40Hz-20kHz) frequency response and superior transient response that promise smooth and natural-sounding vocals. A cardioid polar pattern helps improve rejection of unwanted sounds and combat feedback, whilst a multi-stage grille is said to offer excellent protection against plosives and sibilance without comprimising high-frequency clarity.

£ £99 including VAT
W www.audio-technica.com/en-gb/at2010

Audix VX5

There are currently two handheld capacitor designs in the Audix range. The first, the VX5, has been designed for stage, studio and broadcast applications, boasting a controlled supercardioid pattern that helps to isolate vocals in noisy environments whilst avoiding feedback, and a smooth 40Hz-16.5kHz frequency response for a natural, studio-quality sound.

£ £249 including VAT
W www.soundonsound.com/reviews/audix-vx5-vx10
W www.audixusa.com/products/vx5/

Audix VX10

The second offering in the Audix range improves upon the specifcations of the VX5, and was designed with the aim of setting a new standard for live sound applications. The VX10's high-frequency response extends to 20kHz, and self-noise is improved, too, measuring 19dBA (compared to 26dBA for the VX5). The mic has a cardioid polar pattern with an off-axis rejection figure greater than 20dB.

£ £599 including VAT
W www.soundonsound.com/reviews/audix-vx5-vx10
W www.audixusa.com/products/vx10/

Austrian Audio OC707

Austrian Audio say that the OC707 offers all of the important characteristics of a studio microphone, but can withstand the rigours of everyday stage use. The mic employs the company's OCC7 capsule, which is hand-built at their factory in Vienna before being measured and tuned using an in-house anechoic chamber. It boasts a linear frequency response (35Hz-20kHz), low self-noise figures and is capable of withstanding SPLs up to 150dB, maintaining a natural and open sound even when subjected to extreme volumes. There is also a built-in second-order high-pass filter at 120Hz.

£ £449 including VAT
W austrian.audio/oc707/

DPA d:facto

DPA say that their flagship handheld capacitor model has been designed specifically to deliver the most transparent vocal sound possible. A supercardioid pattern offers high rejection of sounds arriving at the sides and rear of the mic, and also remains consistent across the frequency range to help avoid feedback issues. The d:facto can be purchased with either a Linear (4018VL) or Softboost (4018V) capsule, the latter providing a broad 3dB emphasis at 12kHz that helps vocals to cut through a busy mix.

£ €1056 including VAT
W www.dpamicrophones.com/handheld-microphones#dfactovl

DPA 2028

The 2028 makes DPA's signature natural sound and impressive technical performance available at a more affordable price point. The supercardioid pattern and linear off-axis response remain, and a solid body and grille construction make sure the mic stands up to use in a live environment.

£ €600 including VAT
W www.soundonsound.com/reviews/dpa-2028
W www.dpamicrophones.com/handheld/2028-vocal-mic

Earthworks SR314

Earthworks may be best known amongst live engineers for their high-end drum-mic kits, but their line-up caters to a range of other instruments, too, and includes a pair of handheld vocal mics. The first, the SR314, boasts a tight and consistent cardioid pattern along with a flat 20Hz-30kHz frequency response, and can handle SPLs up to 145dB without introducing any distortion. The mic sports a striking look, and is available in stainless and black finishes, with the latter offering the choice of a stainless or black mesh grille.

£ £699 including VAT
W www.soundonsound.com/reviews/earthworks-sr314
W earthworksaudio.com/vocal-microphones/sr314/

Earthworks SR117

A more recent addition to the Earthworks range is the SR117, which boasts some impressive specifications at a significantly lower price point. The company say that despite the cost, no compromises have been made in the mic's design, which boasts a neutral response and controlled supercardioid polar pattern helping to deliver clear-sounding vocals with the best possible feedback rejection.

£ £238.80 including VAT
W earthworksaudio.com/vocal-microphones/sr117/

»

Ehrlund EHR-H

Like all of Ehrlund's mics, the EHR-H's capsule features a triangular diaphragm that dampens resonances more effectively than traditional circular designs, resulting in an improved transient response and increased transparency at mid and high frequencies. The frequency response features a slight presence lift at around 6kHz, and remarkably, extends all the way up to 87kHz!

£ £625 including VAT
W www.soundonsound.com/reviews/ehrlund-ehr-h
W https://ehrlund.se/product/ehr-h/

Electro-Voice RE420

EV say that the RE420 boasts a rich, detailed and full-bodied sonic signature that is capable of capturing expressive vocal performances in exceptional detail. Its cardioid pattern offers excellent acoustic isolation and feedback rejection properties, but still provides a comfortable working range around the front of the mic. Breath and wind noises are kept at bay by a pop filter design that utilises two dense foam layers with an air gap,and an internal shockmount makes sure that handling noise is kept to a minimum.

£ £309 including VAT
W products.electrovoice.com/la/pt/re420/

Electro-Voice RE520

The RE520 shares many design features with the RE420, but features a supercardioid polar pattern. The same grille, pop-shield and internal shockmount are present, as is the switchable high-pass filter, and the overall sonic character remains much the same. Due to the difference in pattern, maximum rejection is achieved at the rear sides of the mic rather than directly behind, and maximum SPL handling is specified slightly higher, at 150dB.

£ £309 including VAT
W products.electrovoice.com/na/es/re520/

Lewitt MTP W950

Lewitt's flagship MTP W950 underwent extensive beta testing during its development, where it was put to work on the road with some of the world's most high-profile acts. The final design promises premium studio-quality sound, along with impressive resistance to spill and bleed — Lewitt say that the mic offers 90 percent rejection of sound sources to its rear. The windshield design provides protection against plosives and moisture, and the mic boasts a "cupping-safe" design that helps it maintain a natural tone regardless of the user's mic technique. A trio of switches beneath the grille offer a choice of cardioid and supercardioid patterns, along with -12dB pad and 120Hz high-pass filter options.

£ £689 including VAT
W www.soundonsound.com/reviews/lewitt-audio-mtp-w950
W lewitt-audio.com/microphones/mtp-live/mtp-w950

Lewitt MTP 350 CM

The MTP 350 CM has a similar physical design to the MTP W950, but with a reduced set of features at a lower price point. It still offers the sonic benefits of a capacitor capsule, with a frequency response that has been tailored to provide clear and detailed vocals, but with a fixed cardioid pattern and no built-in pad or high-pass filter options.

£ £139 including VAT
W www.lewitt-audio.com/microphones/mtp-live/mtp-350-cm

Microtech Gefell M900 / M910

Microtech Gefell's M900 and M910 are housed within the same body, but offer cardioid and hypercardioid polar pattern options respectively. The capsules employed feature a ceramic backplate, which is more stable in response to temperature changes than the brass designs used by many manufacturers, and there is also an onboard switching power supply which allows the capsule to be polarised at a higher voltage than that of the phantom power supply, resulting in increased sensitivity and reduced self-noise.

£ £1614 including VAT
W www.soundonsound.com/reviews/microtech-gefell-m900
W www.microtechgefell.de/mikrofon?wl=312-M900
W www.microtechgefell.de/mikrofon?wl=313-M910

Miktek PM5

The Miktek PM5 shares the same capsule, electronics and proprietary output transformer as the company's C5 SDC mic, offering a linear frequency response that delivers balanced, natural-sounding vocals. A solid die-cast body construction ensures that the mic stands up to use on stage, whilst a steel grille protects the capsule as well as housing a multi-stage windshield designed to tackle plosives. Although primarily designed for live vocal use, Miktek say the PM5 is perfectly at home in the studio, and can be put to work on sources such as acoustic guitars, pianos and drum overheads.

£ $599
W www.miktekaudio.com/pm5/

Milab LSR-3000

Milab's LSR-3000 takes the cardioid capsule from their best-selling VM-44 SDC and places it in a solid brass body that has been built for a life on the road. The company say that the mic will capture every subtle detail of a vocal performance whilst providing excellent feedback rejection and low susceptibility to handling noise. A pair of switches kit users out with an 80Hz high-pass filter and a -12dB pad, the latter of which allows the LSR-3000 to handle an SPL of 150dB without any audible distortion.

£ £1018 including VAT
W www.milabmic.com/lsr-3000/

Neumann KMS 104 / 105

Neumann's KMS 104 and 105 offer cardioid and supercardioid options, respectively, and aim to bring the sound of the company's renowned studio mics to the stage. Both are equipped with true capacitor capsules and promise to deliver a superior transient response that is said to result in more immediate-sounding vocals than dynamic designs. The design has been optimised for close-miking, and features a fixed high-pass filter to compensate for the proximity effect, as well as being able to withstand SPLs up to 150dB without clipping.

£ £619 including VAT
W www.soundonsound.com/reviews/neumann-kms105
W www.neumann.com/en-en/products/microphones/kms-105/
W www.neumann.com/en-en/products/microphones/kms-104/

Peluso PS-1

The supercardioid Peluso PS-1 comes packed with features that have been implemented with stage use in mind. The capsule is suspended in place by a pair of flexible silicone discs that help to minimise handling noise, and is protected by a rugged metal grille housing a windshield that offers protection against plosives and moisture. Switchable low- and high-pass filters can be engaged to reduce the level of unwanted noise sources, and an 18dB pad allows the mic to accommodate louder sources, helping the PS-1 to withstand SPLs up to 146dB.

£ $746
W https://pelusomicrophonelab.com/ps-1/

Rode M2

Rode's offering comes in the form of the M2, which utilises a half-inch capsule suspended in an internal shockmount, housed in a rugged all-metal body designed to withstand the demands of live use — in fact, Rode are so confident in the design that they offer a 10-year warranty. A tight supercardioid polar pattern helps to reject off-axis sound sources and combat feedback, whilst a 35Hz-20kHz frequency response ensures clear and natural-sounding reproduction of vocals.

£ £116 including VAT
W www.rode.com/en/microphones/live/m2

Sennheiser e865

Combining the rugged design of Sennheiser's 800-series range with the extended frequency response and improved transient performance of capacitor-based designs, the e865 promises studio-quality sound in even the most demanding live situations. Its supercardioid pattern helps to combat spill and feedback issues, whilst the 40Hz-20kHz frequency response is said to offer natural vocal reproduction. Consistency throughout the polar pattern also allows the mic to deliver good results when used at varying distances, or with sources moving slightly off-axis.

£ £219 including VAT
W sennheiser.com/live-vocal-microphone-dynamic-super-cardioid-condenser-e-865

Shure SM86

Shure may be best-known for their dynamic microphones, but their range of stage-focused models now includes a selection of capacitor models, too. One such example is the SM86, which promises to provide live users with studio-quality sound on stage. It features a cardioid polar pattern, and a frequency response that extends from 15Hz to 18kHz with a curve that has been tailored specifically for vocals. The construction is as robust as you'd expect from a Shure SM model, and a built-in three-point shockmount system helps to minimise handling noise.

£ £165 including VAT
W shure.com/en-GB/products/microphones/sm86?variant=SM86

Shure Beta 87C / 87A

Shure's Beta series offers a range of premium designs, many of which are upgraded versions of their popular SM-series models. Although the SM87A is now discontinued, the company still offer the Beta 87C and 87A, a pair of capacitor mics aimed purely at vocal duties. The only difference between the two lies in the polar pattern: the former is cardioid, and the latter is supercardioid. Both offer a wide frequency response that exhibits a low-frequency roll-off to compensate for the proximity effect and a slight 'presence' rise to help vocals cut through a mix, whilst a consistent polar pattern ensures minimal coloration of off-axis sources.

£ £339 including VAT
W shure.com/en-GB/products/microphones/beta_87c?variant=BETA87C

Crookwood Stereo VU Meter

Analogue Level Meter

It's hard to believe that the humble VU meter has been with us for almost 85 years. It was conceived in 1939 by a partnership of American broadcasters (CBS, NBC and Bell Labs) as the Standard Volume Indicator (SVI). It's been used, albeit mostly in simplified and less accurate forms, ever since, and history has proven its practical effectiveness as a reliable indicator of the perceived volume of an audio signal. (VU meters are frequently rejected as being 'old-fashioned', and the BBC long insisted that VU stood for 'Virtually Useless', but neither claim is true: when correctly calibrated, VU meters are remarkably useful, and it's no accident that the Momentary Loudness mode in the BS.1770 Loudness Normalisation metering standard is almost identical to the classic VU meter spec!)

Of course, our DAW software provides digital sample-peak meters and, increasingly, sophisticated true-peak and loudness metering. Some even offer virtual VU meters. But keeping these in view takes up precious screen space, and if they're not kept in view they can't provide the at-a-glance information needed to discern whether or not a mix is in the right volume ball-park. A high-quality, well-designed mechanical VU meter that can be calibrated easily (a feature most omit!) addresses that need very well indeed (as well as looking cool in any studio!), has the advantage of not consuming screen space, and it doesn't need the user to focus intently on and mentally interpret complex on-screen numbers. Instead, it conveys crucial, instinctive level information, even when only peripherally visible, just through the angles of the needles, conveying all the information required in an instant, and allowing the user to concentrate entirely on what is being heard — which is a really good thing!

Crookwood are a small, high-end audio manufacturer based near Salisbury in the UK, and they build an impressive range of mastering consoles, monitoring controllers, routers, converters and mic preamps, and all of it modular and customisable. We've reviewed various Crookwood products over the years, and I've been using my custom Crookwood mastering console almost every day for over a decade now. Their stereo VU meter is available in two formats, with the meters arranged either in a vertical array (to fit into Crookwood's modular console control panel) or side by side on a standard 2U rackmount panel; I was sent the latter version for review. There's a cost option to have it fitted in an oak case, and they also offer a 1U rackmount version with six smaller VU meters for 5.1 metering.

Sifam's superb Presentor AL39W VU meters are employed in both stereo versions. These meters each have a visible display area of roughly 90 x 40mm and they conform closely to the original SVI ballistic requirements, which is key to their accurate audio metering performance. Crookwood have added their own lighting, with LEDs along the bottom of each meter giving the unit a lovely, warm-yellow glow, which looks fabulous — just like the classic festoon incandescent bulbs traditionally used in vintage consoles, but without the heat or short lifespan!

The rear of the panel is open, but the wiring is neat and well secured, and the audio electronics are enclosed in a simple folded metal box at the centre, carrying a pair of XLR sockets for the inputs and a coaxial DC power inlet. A universal wall-wart power supply is included, with clip-on plugs to suit different international mains outlet standards.

A rotary switch in the centre of the front panel is the only user-control, and it attenuates the input signals in six 3dB steps. The attenuation facility is critically important in the meter's practical usefulness and was central to the original 1939 SVI spec, but it's mostly omitted in modern VU meters, for reasons of both cost and ignorance! At the 0dB setting Crookwood's VU meters are calibrated in the traditional way, such that 0VU is displayed for an input level of +4dBu. In most cases, though, mixes and certainly masters are hotter than that — the attenuation settings effectively shift the meter calibration so that 0VU is shown for nominal input levels of +7, +10, +13, +16, or +19 dBu.

The idea of recalibrating the meter reference level is not new, and an early digital reincarnation came in the form of Bob Katz's popular K-meter system, with its K-20, K-14 and K-12 settings. The K-20 setting is equivalent to Crookwood's 0dB calibration, while the K-14 mode can be replicated with 6dB attenuation, and the K-12 mode with 9dB (not perfectly accurate, but close enough!). To choose an appropriate attenuation level, Crookwood recommend playing pink noise at the desired mix level (eg. -14LUFS for streaming) and adjusting the attenuator so the needles sit around -2VU — close to vertical on the scale. Quick, easy, and surprisingly accurate!

Being able to recalibrate VU meters for different nominal operating and mix levels is what makes this particular design so useful in mixing and mastering, and the classic mechanical moving-coil meter implementation provides an intuitive understanding of when the needles are hitting the 'right spot', even with the quickest of glances. In summary, then, this is a very high-quality, accurate, professional-grade stereo VU meter, with adjustable front-panel attenuator. It's also highly practical, easily configurable, looks the business and is relatively affordable — I'd suggest that it's worth 2U of anyone's rack space!

Hugh Robjohns

£ £438. Optional oak case £138. Prices include VAT.
T Crookwood +44 (0)1672 811 649
E sales@crookwood.com
W https://crookwood.com

Waves Scheps Omni Channel 2

Channel Strip Plug-in

Grammy-winning mixer Andrew Scheps has an eclectic portfolio, extending from Metallica to Adele, and over the years he's been a familiar sight at trade shows, sharing his mixing techniques on the Waves booth. The Scheps Omni Channel 2 plug-in is essentially an all-in-one, five-module channel strip based on the sonic characters of the processors and plug-ins Andrew uses in his studio, offering input coloration, compression, gating, EQ, de-essing and limiting, with a choice of four preamp saturation types and four types of compression. Each module offers a choice of M-S, Stereo and Duo (dual mono) routing, and the plug-in introduces no additional system latency so is suitable for live applications as well as in the studio.

This updated version adds features such as the extra-heavy Crush distortion, and a new 24dB/oct filter mode joins the previous 6, 12, and 18 dB options for

the input section's adjustable high- and low-pass filters. A new resonance control makes it easy to create synth-like filter sweeps on the 12, 18 and 24 dB modes, and can aid sound modification tasks such as adding punch to kick drums and toms or changing the apparent weight of electric guitar tracks. A Thump button adds a gentle low-end resonance, which can be useful for lifting basses or kicks in the mix. Crush works well as a parallel distortion processor for adding brightness and attitude. The parametric EQ combines Mid and Tone parametric stages with two traditional parametric/shelving EQs. There's also a new soft-knee compressor mode alongside the existing VCA, FET, and Opt(ical) variants. Importantly, this new version of the plug-in has the ability to host third-party VST3 plug-ins; the earlier version had the same plug-in insert point but could only host other Waves plug-ins. That opens up a lot more possibilities!

The DS2 dual de-esser works using two independent de-essing filters, each with adjustable bandwidth and shape. These can be used together, and although the channel modules are arranged in a logical order they can be rearranged for specific applications. There are side-chain filters for use with both the internal and external side-chain feeds and, as you might expect, there's a large library of presets created not only by Andrew Scheps but also by other top-tier mix engineers.

While most of us will already have individual plug-ins that can handle saturation, EQ, compression and de-essing, the Scheps Omni Channel 2 has its own character, which reflects the tastes and working methods of Andrew Scheps himself. If you need to add definition, energy or a more forward character to a track or mix then this plug-in has a lot to offer, but it can also handle the gentler side of mixing too. The new Crush and resonant filter options bring in plenty of sound-design potential, while the soft-knee compressor adds a certain completeness to the existing optical, VCA and FET choices. The ability to host third-party VST3 plug-ins via the insert points is a very practical and thoughtful addition, so all in all this revised version of Scheps Omni Channel is very welcome and well worth demo'ing. *Paul White*

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SOUND ON SOUND

Strymon BigSky & Deco

Plug-in Effects

Strymon's reputation for making top-quality digital effects pedals is second to none, but recently they've also made some of their creations available as DAW plug-ins. The first two to market are BigSky and Deco — the former is a multi-engine reverb, and the latter offers tape saturation and a double-tracking effect. They both use code that's ported directly from Strymon's hardware pedals, so you know you're getting the real thing, and while Strymon went to great lengths to ensure the pedals' user interfaces were as intuitive as possible, they've managed to make operation even more straightforward in the plug-ins. All the common plug-in formats, including AAX, are supported across macOS (11.6.4 and higher) and Windows (10 and up), with authorisation via an iLok account (no physical dongle required). By the way, if you're unfamiliar with the hardware versions, it would be well worth checking our January 2014 review of the BigSky (https://sosm.ag/0114-strymon-big-sky) and our Deco review, which appeared in *SOS* October 2015 (https://sosm.ag/1015-strymon-deco), both of which are free to view.

Taking BigSky first, as with the hardware, this plug-in offers a choice of 12 algorithms that cover familiar themes, such as real spaces, plates, springs and reflections, as well as more creative takes on reverb such as Non-linear, Cloud, Swell, Shimmer, Magneto (a multi-head tape echo with added diffusion) and Bloom. The main controls govern algorithm selection, decay, pre-delay, tone, modulation, low end and mix, with a Hold switch that can engage either a freeze or an infinite reverb mode. Smaller controls at the top left of the GUI adjust the overall input and output levels, and there's a generous collection of presets to get you started (you can of course save your own settings too). You can opt to see control values or not by clicking the eye icon.

Switch algorithms, and the lower part of the GUI also changes. For example, run the Reflections algorithm and there's a small overhead view of the room with a draggable sound source cursor, while in Chorale mode you can select from a number of vowel filters and resonance options. If you adjust these algorithm-specific controls during a session, their settings persist if you switch to a different algorithm and back. Note that some of the main controls also take on a slightly different function depending on the algorithm. For example, in the Bloom algorithm the tone control becomes a resonant filter and the modulation control is reconfigured specifically for this algorithm.

The BigSky plug-in makes accessing secondary functions even easier than on the hardware.

The Deco plug-in is all about tape echo, flanging and double tracking, but you can also use it just for the tape emulation.

Deco emulates the sound of analogue tape to produce effects such as echo and flanging, though it's also possible to use just the tape saturation emulation to give tracks that 'recorded to tape' vibe. There's a choice between open reel and cassette emulation, the latter replicating the original ALC auto-limiting circuit found in high-end 1970s cassette decks. There are knobs to control Saturation, Volume, Tone, Low Trim, Blend, Wobble, Auto-Flange Rate and Lag Time, plus a small number of buttons.

Flanging and ADT effects were originally created by using two tape machines carrying copies of the same material, and Deco follows this approach to recreate these vintage techniques. It is the time relationship between the two tape machines that creates effects such as phasing, through-zero flanging, chorus, echo and automatic double tracking. The effects can be free-running or tempo-sync'ed, with an Auto Flange option that recreates a smooth, through-zero flange without the user needing to physically adjust or automate the Lag Time control.

The virtual 'reference' tape machine is the one on the left while the one on the right, the Lag deck, can be polarity-inverted, delayed and modulated with respect to the reference. When working in stereo, Bounce mode causes the right channel of the Lag deck to be polarity-inverted and bounced to the left channel input, resulting in a ping-pong type effect. I can confirm that Deco recreates vintage tape flanging and 'John Lennon' ADT extremely convincingly, and it can also create conventional tape delays of up to 500ms.

Both these Strymon plug-ins are seriously good, and BigSky in particular can add magic to pretty much any instrument or voice, not just guitar. Having these effects available as plug-ins can be beneficial even if you own the hardware versions: you can use multiple instances in a project, it's easier and more effective applying the effects to tracks after your recordings have been edited, and you can automate parameters without having to set up MIDI control. Meanwhile the pedals are easier to plumb into a guitar/pedalboard rig, and will work better on stage. There are free seven-day trials available for both plug-ins... but don't be surprised if at the end of that period you feel you simply have to own them! *Paul White*

£ Big Sky $199. Deco $79.
W www.strymon.net

SOUND ON SOUND

PODCASTS

Drum Science - Part 1

The Science of Drum Sound

Part 1 looks at the principles of drum acoustics and discusses some scientific aspects of different pitches, frequencies and overtones, to help engineers consider how this information might affect the way they record a drum kit.

Drum Science - Part 2

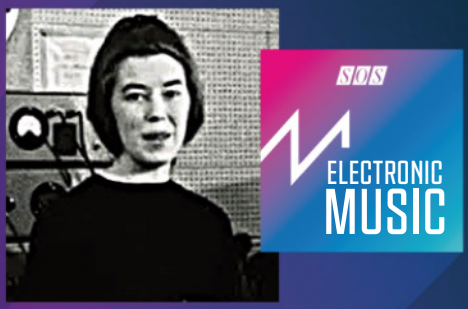
Optimising drums for recording

Rob Toulson explains how to optimise the drum sound for recording by experimenting with drum head, damping system, tuning and drum shell combinations.

NikNak - Turntablism

Turntables In Electronic Music Performance

DJ, Artist and Producer NikNak chats about her route into immersive performances and turntablism, from her time as a student through to winning awards and launching her own albums, EPs and singles with the assistance of various funding organisations.



The Delia Derbyshire Archive

British Electronic Music Pioneer

In celebration of Delia Derbyshire Day 2023 and the 60th Anniversary of the Doctor Who theme, Caro C is joined by fellow devotees Mark Ayres, David Butler and Cosey Fanni Tutti to discuss the Delia Derbyshire archives and the importance of her contributions to the development of electronic music.

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SAMPLE LIBRARIES

Strezov Sampling
Afflatus Chapter II: Brass
Kontakt Instrument

★★★★

The tonal characteristics of orchestral brass instruments are notoriously difficult to capture and reproduce in sampled form. Any library wishing to encompass the full gamut of articulation and tonal colour is going to have to be fairly large, which makes Afflatus Chapter II : Brass appear quite sensible, at an installed weight of 117GB. The remit, as expressed by George Strezov, is to provide a library that can handle contemporary scoring, with reverence toward the great scores from the big screen. This also explains the multi-preset aspect of some of the patching, with references to 'Imperial Brass' and 'Barbarian Brass'.

Away from these multi-presets, the brass catalogue is extensive. Beginning with the French horn section, capture has been undertaken at the solo, two-player and four-player levels. This follows symphony orchestra convention, providing a bewildering array of articulations, tonguings and instrumental colours, both muted and senza sord. The dynamic range of the sampled set glides from subtle pianissimo to an outright onslaught, should you need to appropriately reflect any on-screen nastiness. One revelation, which has made its way across from the Afflatus String package, is polyphonic legato; this is not only a huge timesaver when producing a performance in your DAW, but also incredibly convincing. Even across relatively large interval movements, it really does sound authentic, and is included alongside the more traditional Sustain and Mono Legato modes, with mode switching available from the patch panel.

Also included in the library are cinematic favourites like the cimbasso and bass trombone, as well as a highly convincing tuba section, which just leaves the trumpet and trombone sections. These are provided in solo and three-player units, with metal and useful carbon mute capture, should you crave a softer con sord colour. The three-player units also receive the polyphonic legato treatment, mirroring the horns, though there are some areas of sonic weakness in these sections. Directional brass instruments, with their brighter tone, don't always fare so well with close miking, and certainly with my trombone-player's ears, I would suggest that backing off the close mic signal affords a better overall colour which sits better with the orchestral palette. It also lessens the presence of the occasional tonal blemish, which seems to be present on some louder samples.

As a package, Afflatus Brass is certainly comprehensive, but some sections arguably require a degree of listening and adjustment to get the best from them. The dynamic control is also exceptional throughout, but even hosted on an up-to-date Mac, I had to back off the sample buffer size to 256 samples to prevent audible glitching while altering dynamic range in real time. Afflatus Brass is clearly a powerful and consuming beast, in more ways than one! *Dave Gale*

€658.80

www.strezov-sampling.com

Fracture Sounds
Blueprint: Electric Keys
Kontakt Instrument

★★★★

Fracture Sounds (FS from here on) recently announced their new 'Blueprint' imprint, described by the company as "an ever-growing collection of free sample libraries for Kontakt Player". This first instrument in the series is Electric Keys, based on a vintage Rhodes Mkl Stage 73 Piano. Electric Keys generously incorporates several of the features found in their commercial libraries — a sort of 'lite' introduction to the FS sampleverse, if you will. Blueprint is downloaded and installed via Native Access, thus appearing in Kontakt's Library tab — no hunting around in your file directory to find it is necessary. It also has full NKS support, so works with Komplete Kontrol hardware.

The GUI design is in the established FS house style, clear and self-explanatory — there's absolutely no need for a manual, and indeed there isn't one! All GUI controls can of course be controlled by MIDI via the usual MIDI Learn right-click method. Delving under the hood in the full version of Kontakt, an examination of the key mapping and monitor tab reveal Electric Keys to employ (mostly) four velocity layers, with on average four round robins (one note appears to have as many as 11!), so this is clearly put together with some attention to detail.

The GUI offers four adjustable effects. Stereo Spread: not the usual artificial widening effect, but instead controlling the amount of stereo spread of keys from lowest on the left to highest on the right. Chorus: Kontakt's newer Choral effect, a distinct improvement on the stock chorus of early versions. Atmosphere: a trademark of other FS instruments, this sustaining granular pad layer complements the main Rhodes sound with additional sustain and movement. Reverb: this is editable, with a choice of Hall or Room algorithms plus size, damping and modulation controls. Also included is a choice of Direct or Amp signals, the latter using Kontakt's Twang amp effect fed through a Leslie cabinet impulse response, as opposed to a separate pool of 'amped' samples. Finally, the level of mechanical noises (key presses and key releases) can be individually adjusted to taste.

Tonally, this library tends toward the darker, mellower side — mainly due to its pre-programmed EQ settings. Whilst the top-most velocity layer (116-127) has a pleasing bark, I did have to play reasonably hard to reach the top two layers; perhaps the second loudest layer should come in at a slightly lower velocity value than 101. Tonal consistency across the keyboard is good, with volume well balanced across the entire pitch range.

Owners of the full version of Kontakt will get the most mileage from Electric Keys, being able to delve under the hood to customise the instrument's tone quite radically. Try tweaking the Twang amp settings, or swapping it and the cabinet for different models; in particular, experiment with

your own settings of the EQ found in the Bus 1 insert strip. Blueprint runs in both the free and full versions of Kontakt 7 or higher; Electric Keys itself is a 2.21GB download. *Nick Magnus*

Free

www.fracturesounds.com

Streamtech Music

Iconic Violin

Kontakt Instrument

★★★★

With Iconic Violin, Streamtech Music provide a solo violin recorded in a historic church in Heidelberg, Germany. Presented in the free Kontakt Player, the centrepiece is a fantastic sustain articulation with legato. The portamento style allows for heartstring-pulling slides between notes, and can even be addressed beyond an octave, though the regular legato is more appropriate here. There is no round-robin legato, but the fast legato allows for agile figures, which remain just as agile at *pp* as they do *fff*. The rebow functionality triggers a restatement of the current note, and works fairly well for lyrical melodies, although it can introduce a small change in tone.

The sustain and marcato articulations offer vibrato and non-vibrato, and the violinist's slightly delayed onset of vibrato included in softly triggered sustain and marcato notes is wonderful, and true to a real string performance. The vibrato is also slower to come in at lower dynamics, which further aids realism.

The inclusion of the legato transitions for all of the long techniques is welcome, and allows for much more realistic programming, as the transitions bridge between articulations when switching within a single legato passage. Though the fades between the legato transitions and the selected technique can sometimes be audible, for the most part this works fairly well.

The flautando is the highlight of the decorative articulations. With the violinist providing just the right amount of variation within the samples to create a slightly undulating texture, it works particularly well as a chordal pad.

The short articulations are the standout of the library for me, particularly the spiccatissimos and spiccatos. With a crisp attack, they are tightly and consistently edited throughout the dynamic and pitch ranges, allowing rapid rhythmic passages to be programmed without any timing issues. Very occasionally the odd round robin will stick out in fast passages due to

having a particular natural artefact, however this can be easily tackled using the Round Robins editor on the Advanced page. Combining the varied short articulations allows notes to be accented with detailed, deliberate differences, just as in a real violin performance.

The acoustics of the church in which this library was recorded provide a beautiful reverb tail, with a pleasant tonal balance, however the release tails option on the Advanced tab allows for a surprisingly tight and dry sound on the shorts. With this, I can hear these shorts easily blending with libraries recorded in much smaller rooms. The different microphone options also allow for some good sound-shaping possibilities.

Iconic Violin could easily fit into pop and folk productions, as well as film scores. It provides an extremely polished, well recorded and edited collection of sounds which are intuitive to play, housed in an excellent user interface with some clever tricks up its sleeve. *Thomas Field*

€119

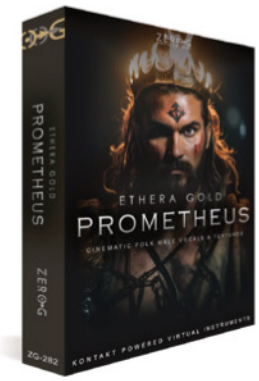
www.streamtechmusic.com

Zero-G

Ethera Gold Prometheus

Kontakt Instrument

★★★★★



Aimed primarily at media composers, Ethera Gold Prometheus is a further collaboration between producer Stefano Maccarelli and Zero-G. *SOS* reviewed the Ethera Gold Atlantis 2 title from the same team back in the August 2023 issue. That Kontakt-based library received a 5-star rating for its combination of epic female cinematic vocal instruments (with some fabulous legato options), vocal phrases, phrase builders, chants, pads and whispers. For Celtic, epic, fantasy or medieval vocal parts suitable for film or game composers, Atlantis 2 is a top-notch choice. And, if you wanted the simplest of comparisons, Prometheus provides a similar musical experience but for the male voice.

Prometheus is an equally comprehensive library. It delivers 9GB of all-new sample content, requires the full version of Kontakt (6.7.1 or later) and provides 38 different Kontakt instruments. There is simply too much ground to cover everything here, but Prometheus includes two different phrase-based instruments with over 5000 phrases in total, both Warrior and Dark vocal chant builders, multiple sustain-based instruments, ritual chants, growls, rhythmic breath options and also a selection of cello, guitar, percussion and sound-design loops.

These are all excellent, but the phrase-based tools deserve a special mention. These are particularly easy to use, letting you construct extended passages where longer musical cues might be required. They are organised into tempo and key-based groups and each preset includes a set of drone, percussion and breath loops mapped above and below the main phrases. It's an instant cue-writing toolkit and a heck of a lot of fun to boot!

However, as with Atlantis, the other star element of the Prometheus show are the various true legato instruments. These come in multiple flavours and, courtesy of the three excellent singers used in the project — Aiden Faith, Davide De Laura and Garmo Mandica — you have plenty of stylistic options under your fingers. So, for example, the three main true legato instruments are labelled Warrior, Dark and Folk and, as the labels suggest, they deliver very different types of performance. Add appropriate ambience effects — and Prometheus has plenty of effects options built into the UI — and the sound is simply glorious.

Ethera Gold Prometheus is a treasure trove of inspiration. It's also a perfect complement to the earlier Ethera Gold Atlantis 2 and, if you have used that library, you will feel right at home here. It is, of course, a specialist title aimed at media composers, but it's also a special one. For dramatic male vocals that would grace any fantasy-style music cue, Ethera Gold Prometheus is absolutely epic. *John Walden*

£90.47

www.zero-g.co.uk

Audio examples of this month's libraries are available at www.soundonsound.com.

DP has everything you need to work with audio for picture.

MATT LAPOINT

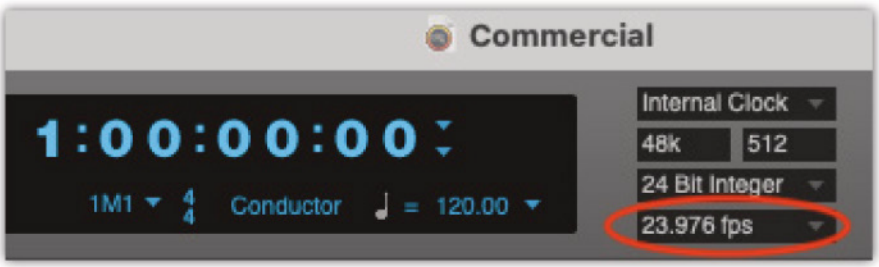
DP's movie window provides all of the workflow features necessary for professional film scoring and post-production audio editing. Let's explore some of the technical aspects of the movie window before examining the settings and features.

A wide range of cross-platform video formats for Macs and PCs is supported, and while MOTU recommend MOV MP4 and M4V files, other standard video stream formats like h.264 will also work. The movie window does not change the video resolution in any way, so standard-definition and HD videos are both fully supported, as are standard and HD frame rates. DP will play back the stereo and multi-channel audio embedded in the movie, as long as the sample rate matches the DP project and the proper bundles exist to route the audio to your audio interface outputs.

Movies can reside in your DP project folders or anywhere else on your local system. It's important to keep track of the location of movie files used in DP projects (perhaps with a clipping or track comment), so that all of the assets can be easily collected when sending projects to clients and/or moving projects to other computers. If the directory changes, a DP prompt (when reopening the project or chunk) will let you choose the new location of the movie file.

When playing back very large, high-resolution video files in a large DP film scoring template file, drive speed, RAM, graphics cards and monitor resolution will impact performance. With 4K videos (and above) it may be necessary to reduce the movie file resolution or obtain a lower-resolution file. When doing so, be sure to maintain the original frame rate.

Screen 1: Choose the frame rate that matches the movie rate here.



In The Frame

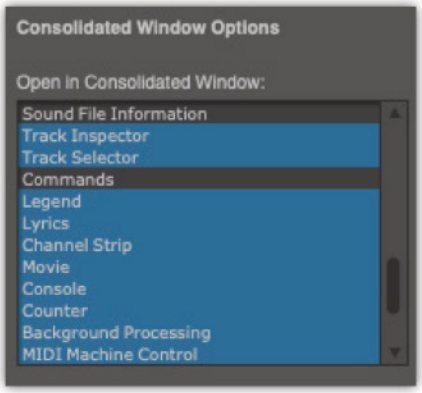
Digital Performer does not auto-detect the frame rate of loaded movies. To determine the movie's frame rate and audio channel format, open the movie file in QuickTime Player and use the Get Info command. On Windows, file properties will list this information.

In DP, the frame rate menu resides in the Control Panel to the right of the counter (Screen 1). Frame rate is a per-project setting in Digital Performer, so you can create and save templates for a specific frame rate that you use often.

To open a movie, choose Project / Movie. The default keyboard shortcut is Shift+V. Navigate to the location of the movie file. As mentioned earlier, it can reside anywhere on your local system. The movie window can either float as a separate window or dock inside the Consolidated window sidebar. To specify the latter, choose Preferences / Consolidated Window Options (Screen 2). In the list, selected windows will dock into the Consolidated window when opened. De-select Movie in the list to make it float as a separate window.

If the movie window is floating, drag the right-bottom corner to resize it. If it is docked in the Consolidated window, resize the Movie window's side bar to resize the movie.

Screen 2: The movie window can either float as a separate window or be docked in the Consolidated Window sidebar.



Movie Window Settings

To access movie window settings, Control-click or right-click anywhere on the movie window. The Set Movie options let you select another movie file from a navigation dialogue. Use Same Movie for All Sequences relates to projects that use more than one chunk. In film scoring, it's useful to use this option if cues are set up as separate chunks that use the same movie file. With this option unticked, each chunk in a project can use a different movie.

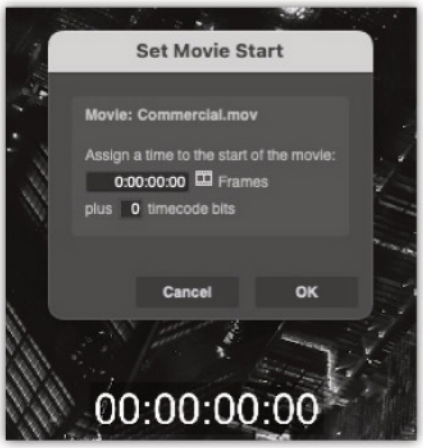
Import Movie Audio imports the movie audio into the DP timeline as an audio track with the same channel format as the movie. After importing the audio, Digital Performer automatically turns down the movie's volume level (in its control panel). Pro tip: always set the movie and chunk start time before importing the movie audio (more on start times below).

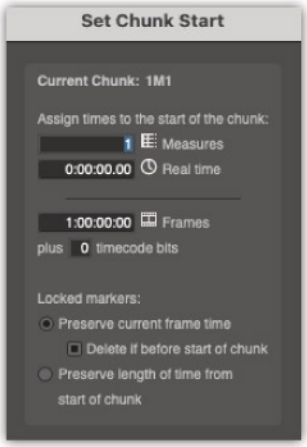
The Lock and Chase options ensure that the movie window syncs, stops and scrubs correctly when rolling DP's transport and editing data. Movie Floats On Top forces the movie window to stay above all other edit windows in DP. In general, it's useful to drag the movie window to another monitor or connected TV to keep the movie window visible at all times while working. When doing so, you can go full screen on the separate monitor (more on this below).

Movie Start Time

Set Movie Start Time is an important setting when scoring to picture. If a timecode (TC) burn-in exists in the

Screen 3: If the movie has a timecode burn-in, go to the first frame and set the movie start time to match it.





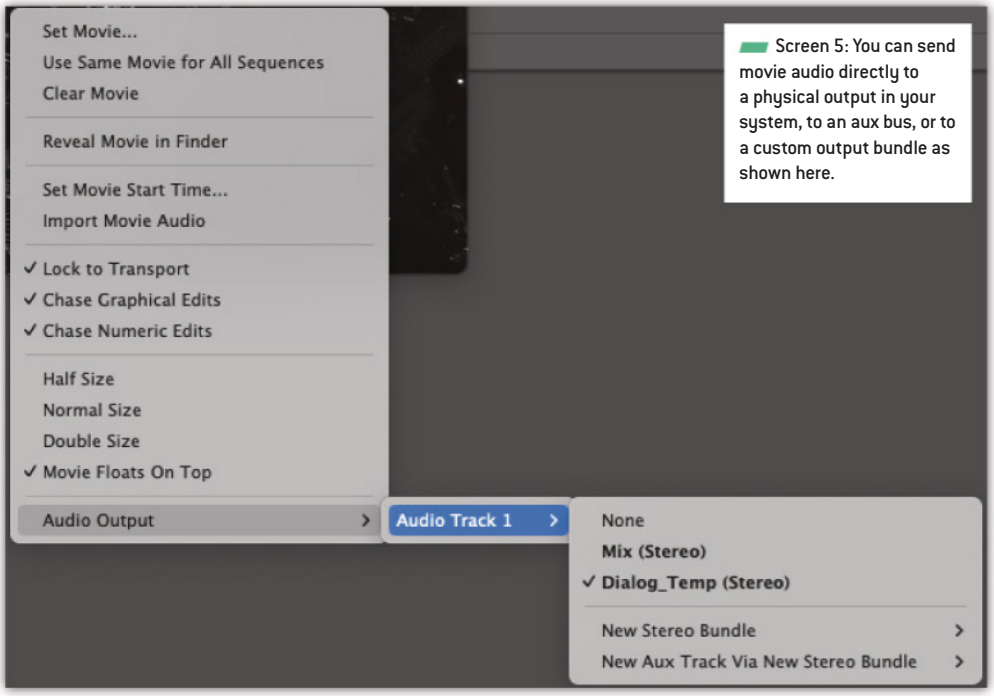
Screen 4: The Set Chunk Start dialogue is accessed in the Chunks window mini-menu. Here you can set the frame start time and the number of the first measure for the sequence.

movie file, use this option to match the movie start time with the first burn-in frame listed (Screen 3). In this example, the movie starts at exactly zero hours.

If the movie file doesn't have a TC burn-in, set the movie start time to the same frame as DP's Chunk Start Time (found in the Chunks window mini-menu). The Chunk Start Time dialogue also lets you choose the desired measure number for the first measure of the chunk (Screen 4), which doesn't have to be measure 1. For example, you might have several back-to-back sequences that are a continuation of each other, where you'd like the start measure of the next sequence to pick up after the last bar of the previous one.

Control Panel

At the bottom of the movie window, the control panel has additional settings. The volume slider controls the audio level. Elapsed and remaining time indicators frame the transport bar. By default, the movie's Play, Stop, Rewind and Fast Forward transport buttons are linked to DP's transport controls. Click the lock icon to unlock them and freely cue/play the movie window separately from DP's transport. On the far right, click the diagonal arrows to go into full-screen mode with the movie. Hit the Escape key to exit full-screen mode. Additional custom key commands can be assigned to movie functions like toggling full screen mode.



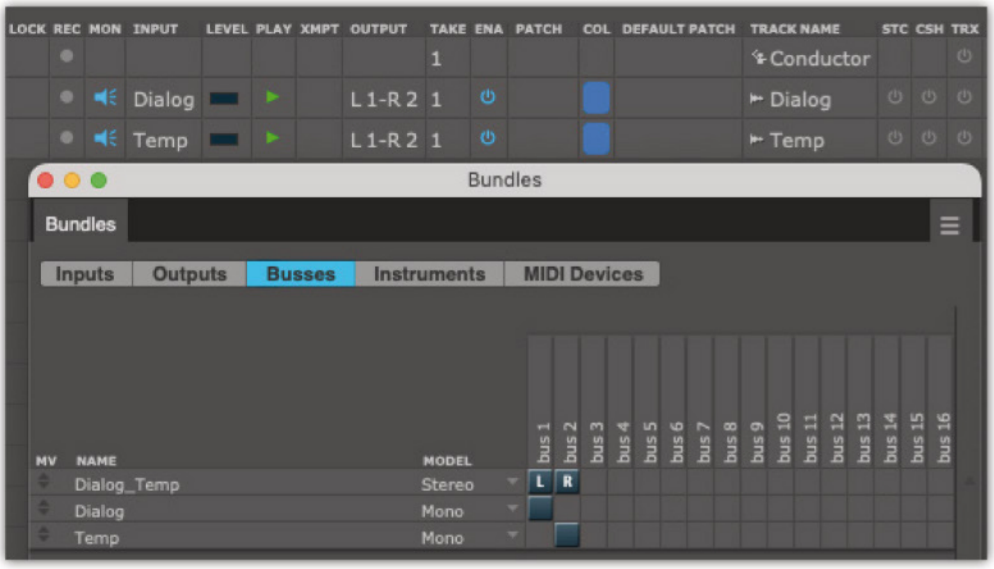
Screen 5: You can send movie audio directly to a physical output in your system, to an aux bus, or to a custom output bundle as shown here.

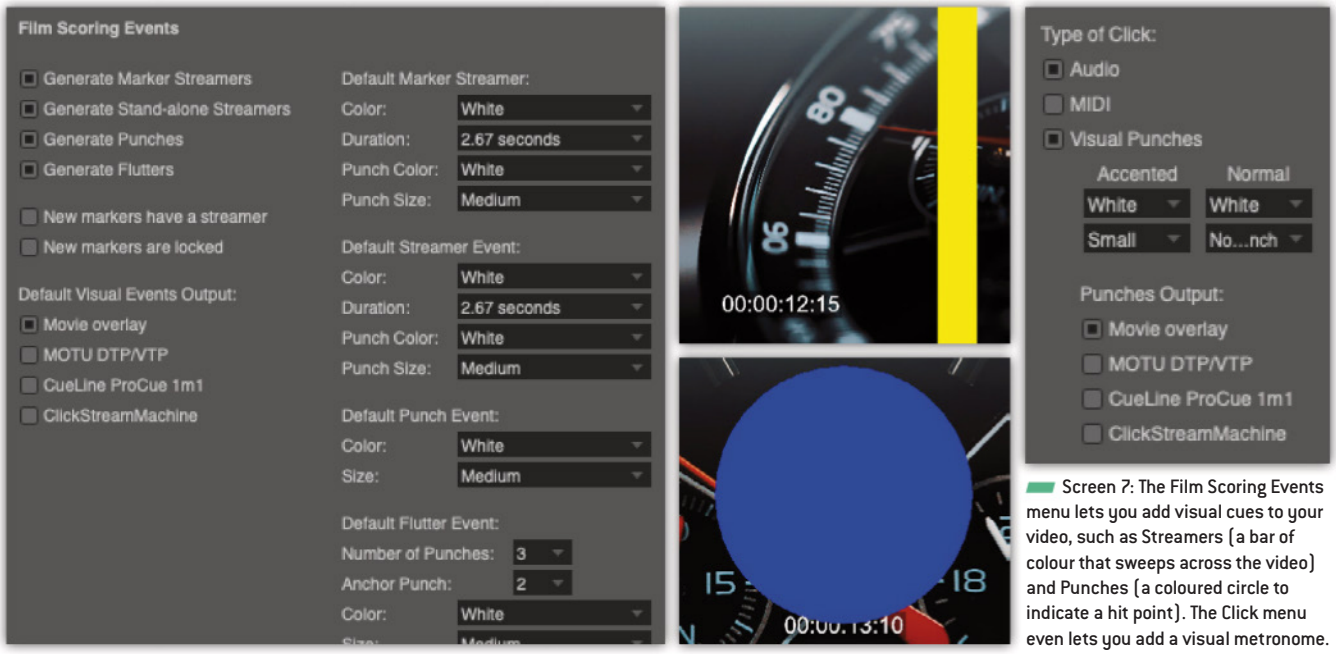
Audio Output

The movie window can send its embedded audio directly to your audio interface on any output pair you select (Screen 5). The output menu also provides custom routings created in the Bundles window. Notice how the movie audio output uses a stereo channel configuration.

You can set up custom bundles for the movie, for example to use separate busses for your dialogue and temp channels, by going to Studio / Bundles. Create a stereo bus and two additional mono busses using the configuration shown in Screen 6. In this example, the movie audio is being routed to the stereo bus pair 'Dialog_Temp' and the separate mono channels at the same time. This is useful when the movie audio puts the dialogue on the left channel of the stereo mix (panned hard left) and temp music on the right channel. Next, create two new aux tracks in DP and assign the inputs to the mono busses and the outputs to the same hardware output. When the »

Screen 6: If the movie audio has temp music in one channel and dialogue in the other, use this bundle configuration to manage them separately when mixing and exporting.





Screen 7: The Film Scoring Events menu lets you add visual cues to your video, such as Streamers (a bar of colour that sweeps across the video) and Punches (a coloured circle to indicate a hit point). The Click menu even lets you add a visual metronome.

» movie plays, the audio is routed to the aux tracks with the proper separation of dialogue and temp music. The aux tracks can then be renamed and rebalanced in the mixer. Use the aux track channel inserts to process the movie audio playback in real time with full automation of plug-in settings, track mute/unmute, level, and so on. To print the dialogue and temp music audio separately, simply replace the mono aux tracks with two mono audio tracks. Assign the same mono bus inputs, arm the tracks, and print the audio in real time.

Alternately, you can accomplish the same thing without aux tracks, by going to choose Studio / Audio Patch Thru and using the Input Only setting. Now, input-monitored audio tracks provide monitoring and printing through the same tracks, without the need for separate aux tracks. Both methods work, and it comes down to personal preference.

Film Scoring Events

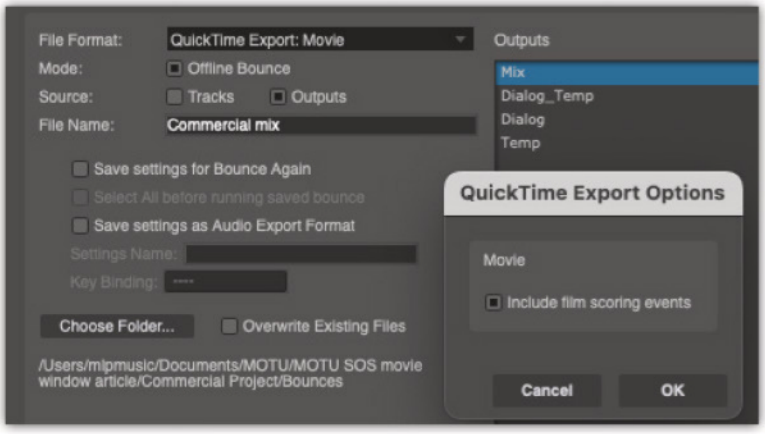
Digital Performer can add Film Scoring Events to overlay on top of the movie. Settings for these events are found in DP Preferences / Film Scoring Events (Screen 7). A Streamer is a vertical bar that sweeps across the movie from left to right to indicate an upcoming hit point or downbeat. Punches are single circular flashes used to indicate hit points. Flutters are a sequence of punched frames that are usually used to indicate downbeats. The Preferences page provides settings for all three types of events. Colour, duration and size are all customisable for each type of event. Film events can also be free-standing events entered into the Event List. They can also be an attribute of markers placed in the timeline, giving you complete control over when and how they occur.

In the Preferences / Click window, you'll find the option to overlay metronome clicks as visual Punches in the movie window. This might be useful in a case where an audible click might be distracting, but a visual click would provide a sense of pulse. Colours and sizes are selectable for both Accented and Normal visual clicks. Visual clicks can be used on their own, or alongside audio and MIDI clicks.

Bouncing (Exporting) A Movie

If you'd like to include the movie audio in your bounced export, print or import the movie audio into the DP timeline beforehand. Then make sure those audio tracks are part of your bounce selection. After selecting the desired tracks and time range for export, choose File / Bounce to Disk. DP will bounce all selected and enabled audio tracks and virtual instruments in the project. The File Format menu provides a variety of compressed and uncompressed audio formats, but to export the movie, choose QuickTime Export: Movie. Select the output, file name and output folder. Once OK'ed, an option to include film scoring events will appear. When checked, Streamers, Punches and Flutters will be superimposed on top of the exported video (Screen 8). The video resolution will match the original file, and the audio will be a compressed AAC file.

The movie window is an integral part of every aspect of Digital Performer, from chunks and songs to clips and QuickScribe, and many other workflows. Its flexibility and frame accuracy fully complement Digital Performer's award-winning scoring features.



Screen 8: When exporting a movie, you can choose to include DP's visual Film Scoring Events.

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Perpetual licence or support plan? Find out which is best for you.

JULIAN RODGERS

Back in March 2022 I looked at the value proposition represented by software subscriptions, and noted the apparent resistance to subscriptions among Pro Tools users online. Online conversations are rarely representative, of course, and a vocal minority can make things appear less nuanced than they are, but the fact remains that there have always been a significant number of people who are wary of signing up for software subscriptions.

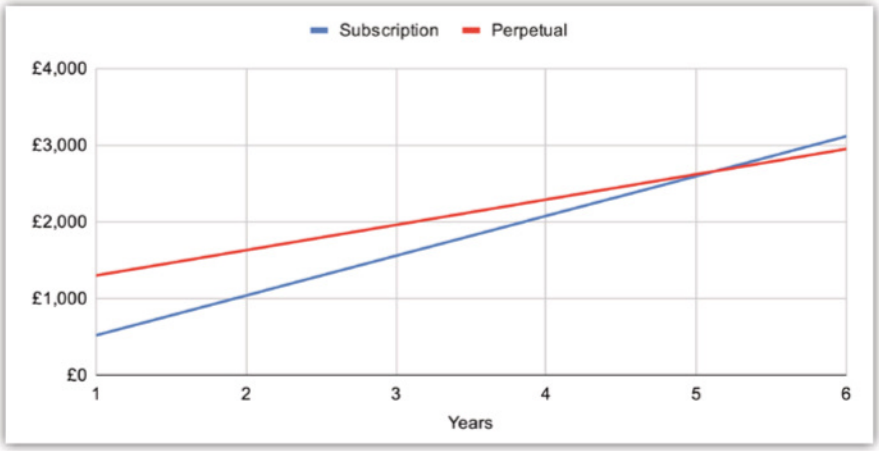
In a surprising move, at the end of September 2023 Avid announced the reintroduction of perpetual licences for Pro Tools, meaning that for the first time since April 2022, it has become possible to buy a new permanent licence. What does this mean and what has changed?

Perpetual Motion

To clarify the situation, perpetual licences have never gone away. Despite some alarmist headlines at the time that Pro Tools had gone 'subscription only', this was never the case. In April 2022 Avid announced that they were ceasing sales of *new* perpetual licences — existing owners of perpetual licences weren't affected by this announcement. If they maintained an update plan they continued to receive all the benefits that carried, and if they chose not to maintain that update plan, their software would continue to work at the version they were on when they chose to go off-plan.

Then an announcement in September stated that new perpetual licences would be reintroduced through resellers (they are not available from the Avid webstore), allowing new customers and existing subscribers to purchase a perpetual licence. Not only that, the offer has been improved. Here's how it works.

When they were last available, Pro Tools Ultimate perpetual licences cost $2599. A Pro Tools Ultimate perpetual licence now costs £1299, a reduction of over 40 percent. An 'off plan' perpetual licence holder who hasn't maintained an upgrade and support plan used to have to pay a Get Current fee. This was Avid's way to encourage perpetual licence holders to maintain an uninterrupted upgrade and support plan. This Get Current charge has been discontinued and there is now a fixed cost to reinstate an upgrade plan. For Ultimate users this cost is £330.



The long-term value offered by a subscription versus a perpetual licence for Pro Tools Ultimate.

This is very significant as it means there is now no financial penalty to be paid for going off-plan, so perpetual licence holders can choose whether to upgrade or not based on the features and functionality being introduced in the updates, rather than to avoid incurring a reinstatement fee. More carrot and less stick, if you will!

For Pro Tools Studio users a similar arrangement exists. The price of a new perpetual licence is £519 and the price of an upgrade for existing perpetual licence holders is £170. It should be remembered that before the reorganisation of the tiers of Pro Tools, the equivalent version of Pro Tools Studio was (confusingly) called Pro Tools, known by many as Pro Tools Standard or even Pro Tools Vanilla to differentiate it from Pro Tools Ultimate, formerly Pro Tools HD.

Thankfully the naming has become simpler. Each tier of Pro Tools has a unique name: Ultimate, Studio, Artist and Intro. The names of the update plans have been rationalised so now for Pro Tools Studio the upgrade product is a 'Pro Tools Studio Perpetual Upgrade' rather than a 'Pro Tools Studio Perpetual Updates and Support Plan Renewal', which was rather a mouthful. Noticing the absence of the word 'support' in the new product I wondered whether this indicated a change to the provision of support, but checking my Avid account I read: "Use a Perpetual Upgrade to renew or get current and receive one year of product updates, support, Inner Circle rewards, Sonic Drop content, and more," so the name and price seem to be the only changes.

Artistic Licence

Pro Tools Artist has always been subscription-only, so unlike with Ultimate and Studio, there isn't any pent-up demand for upgrades from off-plan Artist users. The costs are £170 for an Artist

Going 'off plan' now no longer incurs a Get Current fee if you later decide to renew your subscription. However, you'll still lose access to Avid support, and the new content offered by Avid's Inner Circle and Sonic Drop schemes, until you take out a new subscription.

Perpetual licence and £59 for an upgrade. It's worth knowing that any new perpetual licence comes with a year of upgrades, so there is no need to renew until at least a year after purchase.

There has always been an element of confusion around the difference between a subscription and what was previously referred to as a Perpetual Updates and Support Plan, with many referring to the latter as a 'subscription' because of its recurring annual or monthly payments. However, the key difference is that if a subscription expires, you lose access to your software, just like your Netflix account. With a perpetual licence you always have access to your software. But with this new system of licences and upgrades alongside the subscription offer, it isn't immediately obvious which represents the best value.

The question of which kind of licence offers the best value depends on which tier of Pro Tools you are using and how you use Pro Tools. There are business advantages for professional users to keep a subscription, while occasional users of Pro Tools stand to benefit from the ability to subscribe on a month-by-month basis. Subscriptions are also useful for new customers, as they incur less commitment (ie. up-front cost) compared with buying a perpetual licence. But for long-term Pro Tools users who employ it as their principal DAW, is it cheaper to subscribe or to go perpetual?

"With this new system of licences and upgrades alongside the subscription offer, it isn't immediately obvious which represents the best value."

Like & Subscribe?

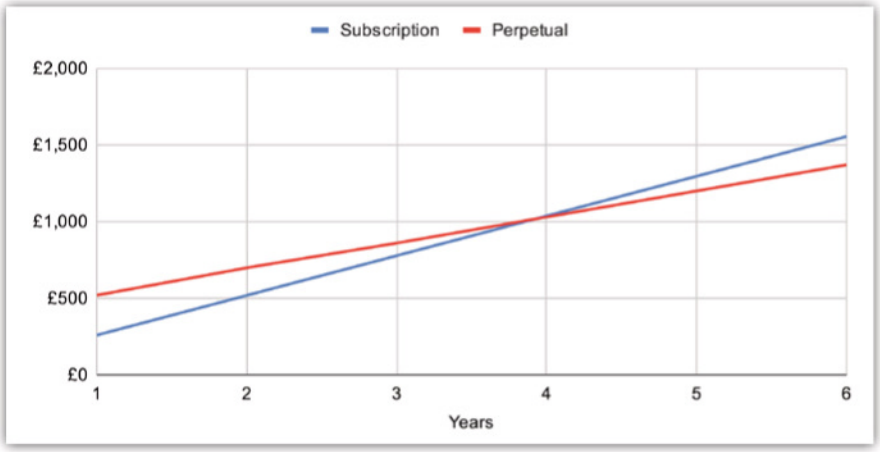
Subscriptions can be paid monthly but the best value is by paying annually. Currently, Ultimate is £519 per year. Comparing like with like, by which I mean both the subscription and the perpetual licence are kept up to date throughout, in the short term it will always be cheaper to hold a subscription because there is no initial purchase of a licence required. However, it's reasonable to assume that at some point the lower cost of an update compared to a subscription will make perpetual licensing more economical in the longer term.

To calculate the break-even period — the number of years it would be necessary to hold a perpetual licence for it to become cheaper than a subscription — it's necessary to calculate the year-on-year costs. Each year a Pro Tools Ultimate subscription paid up-front costs £519. In the first year of a perpetual licence the cost would be £1299. Assuming no price rises, each year the cost of maintaining an up-to-date perpetual licence would add £330, whereas the subscription would continue to cost £519 per year.

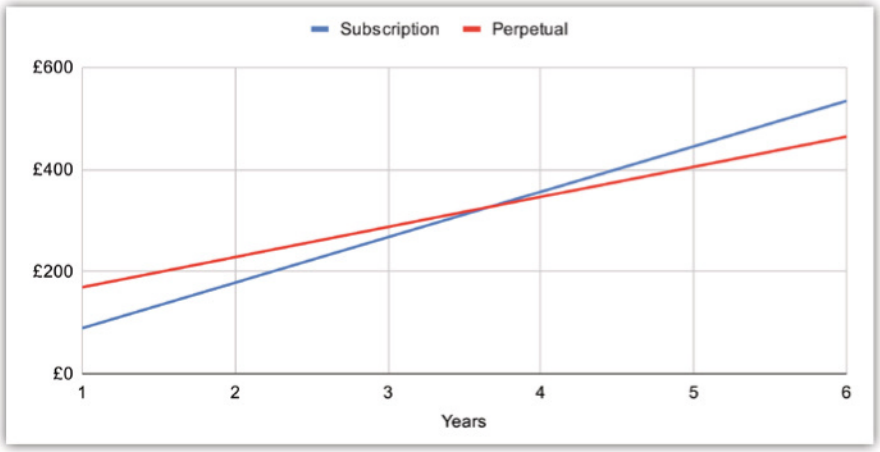
A similar calculation for Pro Tools Studio would be £259 per year for a subscription and £519 in the first year for a perpetual licence, and a cost of £170 per year in upgrades to keep it up to date. For Pro Tools Artist costs would be £89 per year for subscription versus £169 for a perpetual licence, followed by £59 per year for updates.

The graphs show, then, that the break-even point for Ultimate is five and a half years. For Studio it is four years, and for Artist users it occurs during the third year. However this is making the assumption that a perpetual licence update would be maintained throughout this period. Users who go off-plan will lose access to some of the benefits that come with being on an active update plan. Perhaps the most significant of these is the support element, but also plug-ins and the rolling monthly offers and content provided by the Inner Circle and Sonic Drop programmes.

With the removal of the Get Current reinstatement fee there is no longer a financial penalty for users who choose to spend time off-plan. This is a very positive statement from Avid, and suggests a lot of confidence on their part that the new features which will be coming in future updates will be compelling enough to make holders of perpetual licences want to stay current. Less stick and more carrot indeed!



A Pro Tools Studio subscription becomes more economical than a perpetual licence after about four years, assuming the perpetual-licence holder pays the annual upgrade price.



For Artist users, the break-even point is between three and four years.

Wrestling with low-mid mud? Cubase's dynamic EQ can help.

JOHN WALDEN

Almost every instrument group in modern music productions can contain significant energy in the low mids (broadly speaking, around 250-500 Hz), so when mixing you'll often find that there's an unhelpful build-up of energy within this range — if there's too much going on in this region, a mix might sound 'muddy', 'muffled' or 'boxy', and you need a plan to address that.

There are a number of ways in which this 'too much mud' issue might be approached, and careful arrangement and instrumentation choices can obviously play a big part, by which I mean you can write the parts and pick and layer your sound sources in a way that avoids an unhealthy build-up of energy in the low midrange in the first place. But if further control is required, standard EQ applied at the master bus, on subgroups, at the individual instrument level or a combination of all three can be used to control the energy that sits in this frequency range. Cubase offers several EQ tools that can be used for that, including the Channel Strip EQ, StudioEQ and, for users of Cubase Pro or Nuendo only, Frequency 2.

In addition to normal EQ duties, Frequency 2 provides an additional level of control: dynamic EQ. Compared to a standard EQ, where any cut (or boost) is applied all the time, with a dynamic EQ, the cut (or boost) is only applied when the signal is at its loudest in the target frequency range. There are plenty of third-party options for those using other versions of Cubase (such as Tokyo Dawn's freeware Nova: https://www.tokyodawn.net/tdr-nova), but since Frequency 2 comes with Cubase Pro and its GUI makes everything really easy to configure, let's use it to explore some dynamic mud management tactics. I've also prepared some short audio examples (available on the *SOS* website at https://sosm.ag/cubase-1223) to illustrate what I'm discussing here.

Mud Master

If you think the correction required is a modest final tweak to an otherwise finished mix, or you're already at the mastering stage of a project, some corrective low-mid EQ on your master bus or stereo mix might be the most appropriate, or perhaps even the only option. But even if not, I find it's often worth trying an instance of Frequency 2 on the master bus before you move on to a more finessed approach. You might be surprised by what you can achieve!

With Frequency 2 inserted on the master bus, you can apply gain reduction only to the low-mid frequencies, and only when they contain the most energy — this can be an effective tactic if faced with a muddy-sounding mix and static EQ can't get things where you want.

The first screenshot shows the settings I configured for this approach with the mix used in the audio examples. Note that I've switched the Frequency 2 display to Sing (single band) mode, as this provides easier access to the full range of EQ controls for a single EQ band (in this case, band three of the eight available bands), including the Dynamics section. The required settings will be dependent upon the source material, of course, but there are some useful guiding principles. I've used a peak filter centred at 350Hz and with a Q of 2.5 to deliver 9dB of gain reduction. No, that's not a mistake! A 9dB cut on the master bus would almost certainly be too much with a conventional static EQ band; as can be heard in the audio examples, it would rob this mix of too much low-mid energy, making the mix sound 'thin'. But this won't be a conventional EQ cut...

Before we engage the Dynamics section, though, there are a couple of points worth noting. First, with this combination of settings, the filter shape means the cut extends a little beyond the 250-500 Hz range. You can control this by adjusting the Q (larger values make the filter narrower) or gain (smaller cuts also reduce the range of frequencies affected by the filter). Second, when you know you'll be making the band dynamic, it's actually not such a bad idea to start by applying a static EQ that produces rather more cut than you think you'll eventually need. This makes it easier to hear the EQ change, and I find it helps when targeting the initial Freq and Q settings; it'll be that bit easier to hear exactly where the problem lies.

Get Dynamic

Having brought our 'mud spot' into focus, we can now engage the Dynamics section. The main control options are Threshold, Ratio, Attack and Release, and will be familiar to anyone who's used a conventional compressor before. Again, the settings could vary according to the material, but you probably don't want the gain reduction to be too aggressive. In the example, I opted for a gentle 2:1 ratio, fast (3ms) attack and medium (150ms) release. With those set up, I simply adjusted the Threshold fader, to specify the signal level at which this band-specific compression started delivering the amount of gain reduction that my ears suggested was required.

The GUI provides some very useful visual feedback, and it's worth noting that the original Freq, Q and Gain settings control the absolute maximum amount

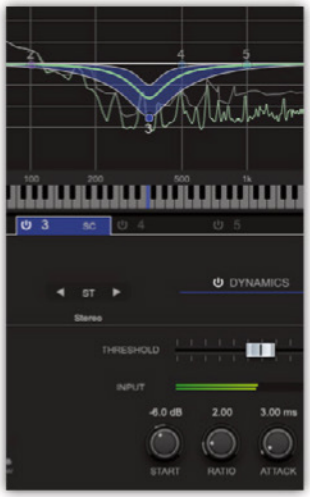
of gain reduction that can be applied, while our Dynamics settings control how often and how far the gain reduction moves towards that maximum position. When there's more 'mud' — that is, more energy within the specified frequency band — more gain reduction is applied so, almost as if by magic, the unwanted mud is dynamically managed. You can hear the result in an audio example on the *SOS* website for which I applied different Threshold settings to adjust the level of mud reduction. The degree of control is impressive and if you get the balance right, you can tackle the mud without making your mix sound thin.

Before we move on, a final thing to note about the Dynamics section is the Start control. This allows you to combine your dynamic EQ adjustment with a static cut/boost. So, for example, if I set Start to -3dB, I would get a static 3dB gain reduction applied all the time and this would then form the starting point (hence 'Start') from which any further dynamic gain adjustment would applied. Again, the frequency display in the upper panel provides a very intuitive graphical representation of this so it's easy to see what's happening.

Group Therapy

If your usual approach to mixing makes use of the group channels to provide separate sub-mixes of the drums, bass, guitars, keyboards, vocals and so forth, then you can apply precisely the same basic principles at the subgroup level. Simply insert an instance of Frequency 2 on each of group channel and use each one to dynamically control the lower mids of each instrument group.

In principle, this should give you greater control over how and where your mud management is applied, and minimise collateral damage to sources that you don't want cut. For example, you could use the approach on just your guitar and keyboard/synth busses (common culprits in mud supplies), while avoiding stealing any lower-mid energy from other instrument groups. Equally, since you're dealing with single instrument groups, you can generally be a little more conservative (for example, a higher threshold or lower ratio). Cuts are only applied to a specific instrument when that instrument exceeds the threshold, and when combined with similar cuts being applied in isolation to other channels, the overall result should, hopefully, be a mix with a little more clarity.



The Start control lets you combine static and dynamic EQ. I've exaggerated the settings here for clarity: the Start setting of -6dB (topmost thin white curve) forms the EQ starting point onto which any further dynamic cuts are superimposed.

By the way, there's nothing to stop you combining both the master bus and group bus approaches, although if the main mud management is done at the group level, you'll find you can apply much more subtle settings at the master bus, just to catch the occasional 1 or 2 dB of excess mud.

Keeping Your Balance

So, where fader moves and conventional compression and static EQ boosts don't seem to deliver a satisfying tonal balance (it's rarely a bad idea to try such bread-and-butter tactics first!), looking to these Frequency 2 tactics to minimise the mud can be a great next step. Of course, you could extend the same basic approach down to the individual instrument channel level, though do use your ears to decide if its actually needed — there's no point processing audio for the sake of it, and if there are complex phase relationships between different sources you should listen out for unwanted side-effects; unpicking problems you only notice later can be time-consuming and frustrating! Another tactic is to use Frequency 2's external side-chain input options to duck one or more sources out of the way of a source whose low-mid presence you want to retain (for example, using the guitar bus to duck a synth bus), but that's perhaps a topic for another day.

Applying Frequency 2's dynamic EQ to subgroup tracks gives you greater control over which instruments are 'de-mudified'.

Studio One now includes some powerful notation and tablature options.

ROBIN VINCENT

Our expectations of score writing within DAWs can be very mixed. Some programs ignore it completely, whereas others, such as Cubase, have had it built in for years. Another approach is to forge a connection between two platforms, as Avid did with Pro Tools and Sibelius. PreSonus have their own fully fledged scoring software called Notion, and there are facilities whereby you can transfer MIDI tracks back and forth, pull audio from one to the other or run Notion as a ReWire instrument within Studio One. It works well, and for professional scoring Notion is a great way to stay in a fully connected DAW environment while working on the dots.

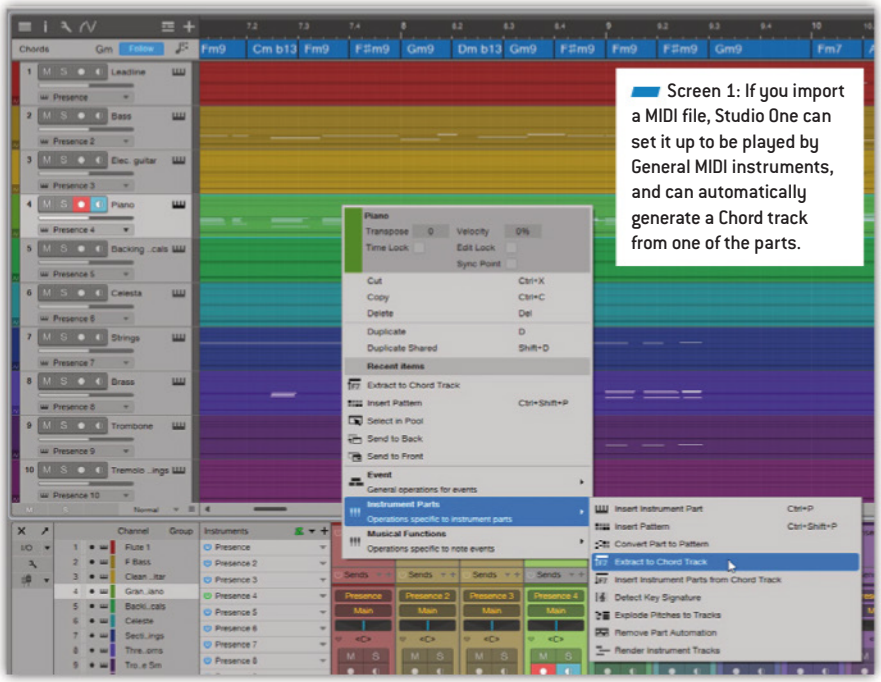
However, ever since PreSonus put basic scoring tools into Studio One version 5, more and more features seem to be making it across the Notion-Studio One bridge. With Studio One 6.5, it's the guitarists' turn to feel the warmth of some attention in the scoring, chord sheet and tablature department.

Strike A Chord

PreSonus added support for basic tablature in version 5.2 but forgot to include any ability to add chord letters or symbols. It was a little frustrating; however, now you can. In fact, the flow of chords through Studio One is so fluid that you'll instantly forget that it wasn't there before.

It all goes via the Chord track, and an example might help show you how well it flows. I pulled in a free MIDI file of the ageless Barry Manilow song 'Copacabana'. Studio One helpfully asks if you'd like to use some General MIDI instruments to play it, and then spreads the parts onto individual tracks. If you find a track that mostly plays chords, you can ask Studio One to extract them to the Chord track. To do this, right-click on the event, go down to Instrument Parts, and find Extract to Chord Track (Screen 1). The chords then appear along the top of the arrange window. Studio One does this pretty flawlessly, although you might find the odd gap when the piano part wasn't playing. You can, of course edit the chords to your satisfaction.

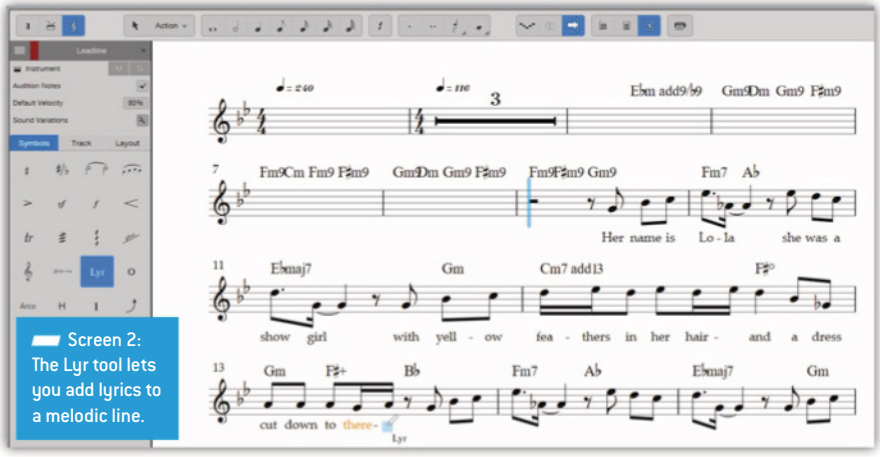
Now we have these chords, we can create a lead sheet with the melody line and chords so you can busk 'Copacabana' down your local tube station or marketplace. Double-click the melody line track, and the MIDI editor will open in the bottom half of the screen. Select the Score View from the toolbar, and you should see the melody written out for you. I find that Studio One doesn't always get the key right automatically. You can set it in the Chord track inspector window (click the 'i' button) or on the transport bar to the left of the tempo.

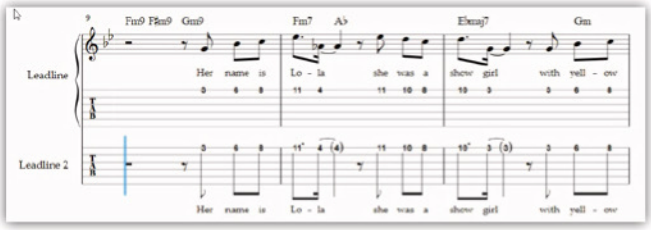


Screen 1: If you import a MIDI file, Studio One can set it up to be played by General MIDI instruments, and can automatically generate a Chord track from one of the parts.

So, where are the chords? On the left of the score is the information panel. There are three tabs: Symbols, Track and Layout. Select Track and tick the Show Chords box. And there they are, pulled from the Chord track like it was no bother at all. The chords are sync'ed to each other, so that if you edit one in the Chord track, it will change in the score. And that's a good point to mention, because you can't actually edit the chord symbols on the score itself; you can only edit them in the Chord track. It feels like the sort of function that could make it into a future update, because going back and forth from score to Chord track is not a very efficient way of handling it.

To see the piece as a sheet of music, click on the Single Track Page Layout button next to the printer icon. When it comes to the layout and printing of your chord-infused music, you don't have much control over how the chords look. You can't select them, edit them, move them or resize them in the Score View. However, you can make some adjustments under the Layout tab by altering the Staff Spacing if your chords are a bit too cramped. Adding lyrics can help space the bars out and give the chord symbols more room as Studio One automatically adjusts to fit the words



Screen 2: The Lyr tool lets you add lyrics to a melodic line.

Screen 3: A lead part rendered in Standard + Tab mode (top), and again in Tablature mode.

sensibly on the page. You can add lyrics by using the Lyr lyrics tool under the Symbols tab (Screen 2). Simply click on the first note with the tool and start typing. Use Space or Tab to jump to the next note, and use hyphens to break into syllables. All the lyrics also appear up in the Lyric track in a nicely connected way.

One thing you can do with the chords independently in the score is transpose them. So, if you want to play with a capo, you can use the Transposition feature under the Track tab to change the key without affecting the Chord track or playback of the song. It would be nice to see a dedicated capo feature in there at some point.

Tabbing Out

Tablature has been in Studio One since version 5.2, but now it's a bit cleverer and comes with a whole raft of symbols for a more nuanced and expressive transcript. For the uninitiated, tablature shows lines representing the strings on the guitar, and numbers that represent the fret position for your fingers, giving you the pitch.

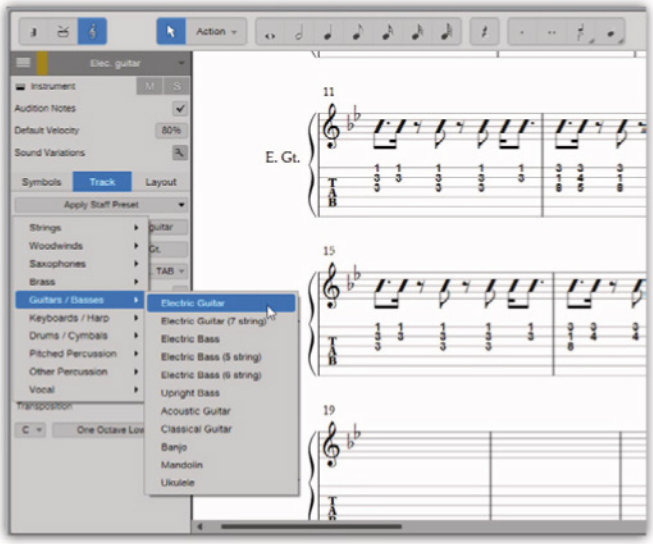
Depending on your chosen staff type, there are a couple of different ways of viewing tablature. Under the Track tab, go down to Staff Type and you'll find two tablature choices: Tablature and Standard + Tab. With the Tablature option, the musical staff is completely replaced by a tablature staff. All the notes are translated across to appropriate strings, and they carry with them rhythmic stems to show how the notes are played. With Standard + Tab, the tablature staff is attached below the music staff, but all of the rhythmic qualities are retained by the notes above, leaving the fret numbers alone on the tab. You can, however, turn the stems on or off.

In Screen 3 I've duplicated the lead part and set the first to Standard + Tab and the second to Tablature so you can see the difference.

Note editing is a little bit different in tab. If you click on a standard note, you can drag it up and down on the staff to another pitch. On the tab staff this results in moving the note to another string, but keeps the same pitch and changes the fret number. This is actually really useful because Studio One isn't going to know which string you want to play each note on, so you'll probably spend a lot of time moving the notes to the right one. To change the pitch, you can either select the fret number and then use the arrow keys on your keyboard to move the pitch up and down, or double-click the number and enter a new one. To add notes as fret numbers, you first select the length of note you want from the toolbar and then click on the string. A text box will appear for you to enter the fret number.

What if you'd like a different sort of tuning, or perhaps you're playing a differently stringed instrument? Under the Track tab, there's a Tab Type option (Screen 4) that includes lots of different guitar tuning types as well as options for mandolin, bass, banjo and ukulele. If that still doesn't cut it, you can manually specify the string values under Tab Strings.

Along with the basic tab functionality, you've also got a bunch of symbols you can add to describe the way you'd like your music played. You'll find these at the bottom of the Symbols list. Under the



Screen 4: There are tab presets available for a range of stringed instruments.

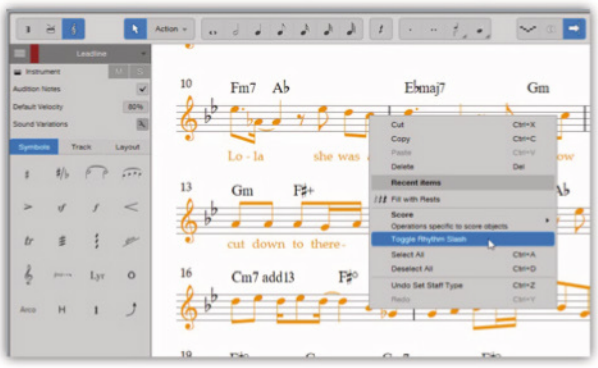
Bend icon, you've got 13 different sorts of pitch modulation, which include all sorts of ways of wobbling the whammy bar and whether you bend, pre-bend, hold, release or do everything all at once. You'll also find hammer-ons, pull-offs, tapping and plucking symbols; it's pretty comprehensive. However, none of these graphical niceties translate into actual expressive control over the MIDI instrument playback in Studio One; they are just for the printout.

And finally, another way of enabling tablature is through the Staff Preset function. If I select the electric guitar track from my 'Copacabana' MIDI file, Studio One is bright enough to automatically interpret the General MIDI sound selection into providing guitar tab in the Score view. But you can do this for any track, regardless of name or intention. Under Track and Apply Staff Preset, you have a number of ready-made guitar presets that will do all the heavy lifting for you.

If you want to print out music for guitar players, Studio One has you very well covered. It will be interesting to see if PreSonus take it further in respect to passing the expression marks from the score back to the project.

Rhythm Slashes

If you would like to give a chord sheet to a guitarist more interested in rhythm than melody, you can present the score using rhythm slashes instead. To do this, it's best to select all the notes first using Cmd or Ctrl+A and then right-click any note and click Toggle Rhythm Slash from the menu. The melody line might not be the ideal rhythm for the guitarist to follow, but that gives you an idea of how to do it.



Enabling rhythm slashes on a chord sheet.

Q Do I really need an RIAA preamp to record vinyl?

I want to record some vinyl into my DAW, and I know that I need not just a turntable but also an RIAA preamp because this applies a de-emphasis EQ curve. But does this preamp serve any special purpose other than that — as in, is it also performing any level boost or impedance matching functions? Or, given that I'm recording rather than playing back vinyl records, could a turntable simply be plugged into any audio interface and the EQ applied after amplification?
Cam Levy via email

Hugh Robjohns, Technical Editor

Yes, as well as the slightly complicated de-emphasis filtering, there's impedance matching and lots of gain involved. The impedance matching is critical to the resulting frequency response as well as the dynamics, and there are very different requirements for MM (moving magnet) and MC (moving coil) cartridges. The resistive element is generally fairly consistent for a particular cartridge type, but the capacitive element can vary a lot for MM pickups, depending on the cartridge and the tone-arm cable wiring. MC pickups typically need around 70dB of gain while MM pickups more usually need 40dB or so. Most RIAA preamps use conventional voltage gain configurations, like conventional mic preamps, but moving-coil pickups are really (balanced) current generators and so there are some specialist RIAA preamps that use a current mode for MC pickups that is said to offer better performance. It's possible to connect a turntable directly to some audio interfaces, but not many. RME's ADI-2/4 Pro SE (www.rme-audio.de/adi-2-4-pro-se), for example, can accept, amplify and equalise direct MM cartridge inputs but the impedance-matching options are limited. The tricky thing is optimising the gain structure while applying such enormous EQ correction — leaving enough headroom to accommodate clicks while maintaining a suitably low noise floor. The latest A-D converters are just about there, but it's a bit borderline. For now, I think a good analogue RIAA preamp remains the best quality option.

Vinyl turntables require an RIAA preamp, not only to EQ the signal appropriately, but also to present the right impedance to the cartridge and apply a suitable level boost — few audio interfaces can do this properly.

Photo: Wikimedia/Creative Commons

Q Can you recommend an in-line booster preamp?

I have a pair of Royer R10s and a Peluso SR14, all ribbon mics. I primarily record classical and jazz music, and while the output level from the mics is fine for close jazz recordings they're pretty quiet on classical music, needing the gain on my console preamps to be maxed out, making them a little noisy. So I'd like to purchase an in-line gain booster and would appreciate your recommendation. Also, is it worth paying extra for a variable impedance feature, such as on the Cloudlifter CL-Z?
***SOS* Forum post**

Hugh Robjohns

The low output level of passive ribbon mics is a well-known issue and Royer offer several active ribbons that overcome this, but they've also designed their own in-line gain boosters for use with passive ribbons, to help resolve the kinds of issues you're experiencing. I've reviewed both the single-channel dBooster and the dual-channel dBooster2, and found both to be very sophisticated designs that perform superbly well. For me, a really important feature is that the gain is switchable between 12 and 20 dB. Most in-line gain boosters provide a fixed gain of around 25dB, but I've often found that's too much, risking overload at the console. The dBooster's 12dB option fits the bill perfectly surprisingly often, boosting the signal enough to reduce the workload on the preamp, while minimising self-noise and not risking the headroom. It's true that Royer's sophisticated dBooster2 is more expensive than many rivals' products, but I can recommend it wholeheartedly and suggest that it would be the ideal solution for your particular situation: the highest-quality two-channel unit, designed with Royer ribbons in mind, and enabling different gain structures for different scenarios. The dBooster2 can also serve as a great stereo recording DI box, too, should that prove useful.

As well as offering two channels of in-line gain, Royer's dBooster2 helpfully has two gain settings.

As for input impedance options, the sound of passive ribbons (and moving-coil mics) often changes significantly with different load impedances. Most ribbons deliver their best technical performance when working into higher impedances: over about 3kΩ, and some even prefer loads as high as 18kΩ. For the record, the dBooster presents a fixed input impedance of around 3.6kΩ. Lower impedances act as a kind of dynamic tone control, but it's impossible to predict the effect of any specific settings. While there are a few brands offering adjustable input impedance, the only model I've tested with variable impedance options is the single-channel sE Electronics DM2 TNT. That particular model also happens to offer switchable 15/30 dB gain options, along with a comprehensive set of switched impedances ranging from 50Ω to 10MΩ. If you want the ability to experiment with different impedance options, the DM2 sounds good and would be worth adding to your shortlist, although I found it had less headroom than the Royer model at the low gain setting.

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SOS WHY I LOVE...

SONIC VACATION ALBUMS

SONAL D'SILVA

I've worked in audio for over a decade now, but on vacation I documented things just like everyone else does: with photos and video. That changed after a work trip to the Seychelles. I stood by the side of a dirt road, waiting for the rest of the crew to show up. It was hot and humid, and I was beginning to feel sleepy at 10am, so to keep myself awake, I turned on my Zoom H4n and trained it on the mundane sounds around me: the idling engine of the crew truck, the crunch of gravel as the locals walked by, walla from the fruit stand as people waited for fresh slices of papaya and tangy mango. When I listened back, a few months later, I was surprised at how vivid the memories were — like the audio was a codec that contained a massive file of smells, sights, and other hidden sounds, ready to be uncompressed.

Since then, I've created a sonic album of every trip I've been on and it's glorious; I can't believe I didn't think to do it earlier. I saw the tallest breaking waves I've ever seen on a beach in Sri Lanka but without any real reference for scale, the photo did not do them justice; the sound, though, was pure power. The wind sweeping autumn leaves along the street reminds me how cold it was in the Hague when we got lost and walked in the wrong direction for 20 minutes. I crank up my recordings of a fading campfire in the sleepy hill station of Panchgani, each crackle and pop becoming softer as the flames died down; in the background I hear voices from the bungalow at the top of the hill, arguing about the rules of Uno. The heavy doors of the Paris metro closing with a hiss remind me of how we stood on the platform looking at a poster for a Mark Knopfler gig and wondering if we should go. "Some other time", we said. A few months later, the pandemic happened.

The great thing about creating an audio album is you can now do it with any recording app on your phone. If you'd like to carry audio gear through airport security, do so, by all means, but be prepared for encounters like the one I had where I was asked to explain my Zoom H4n. I showed the security officer how the recorder worked; she stared at me, and picked it up to see for herself. Then she said, "Zap zap!" and waved it about as if she were using a Taser. "That's what it looks like," she said, laughing, as I laughed along, ready to be done with that little interaction.

It's worth all the trouble, though, I realise as I listen to another recording from my Sri Lanka album. Our hotel was located at the edge of a national park and animals regularly wandered the property. A hotel-mandated guide escorted us to our outdoor dinner and I turned my recorder on as we walked in silence. Crickets, our footsteps, and then another sound in the distance. I whisper to our guide, "What is that? Monkeys?" He hesitates, then whispers back, "No, ma'am. Your friends." Indeed, that was the sound of the rest of our party trying, and failing, to keep it down at the poolside dinner. No photo could have captured that moment in all its hilarious glory.

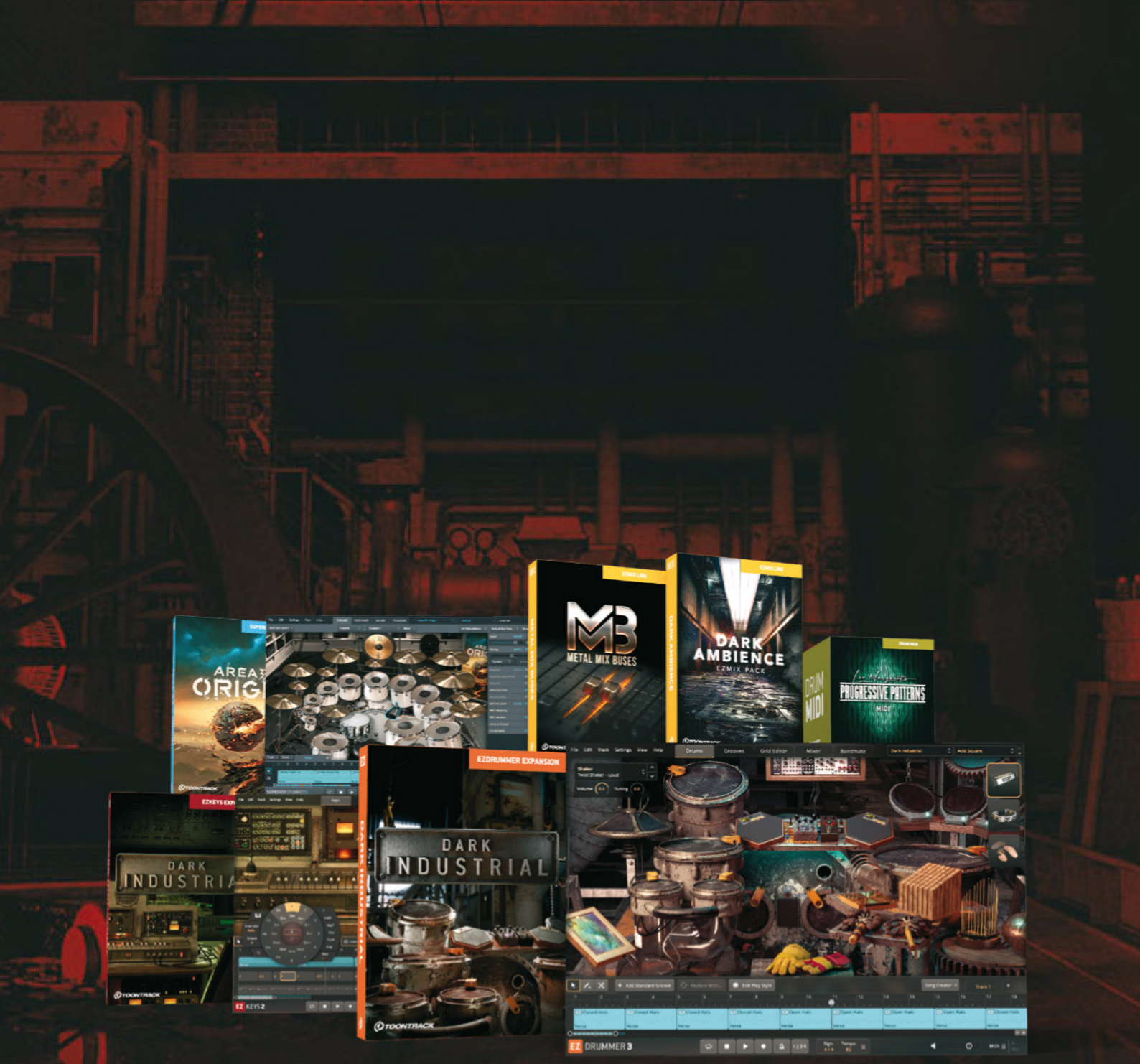
NEXT MONTH IN SOUND ON SOUND

Norwegian Black Metal

In the first part of a new series, we talk to producer Eirik 'Pytten' Hundvin about his unique philosophy and the making of Mayhem's genre-defining album *De Mysteriis Dom. Sathanas*.

February issue on sale **Thursday 18th January**. Available at WH Smith and all good newsagents. Subscribe at **www.soundonsound.com/subscribe**.

Photo: Emerson Posadas



After yet another epic Metal Month, we can add six new releases that span the entire spectrum to our archive – from old-school to hard, heavy and electronic.

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